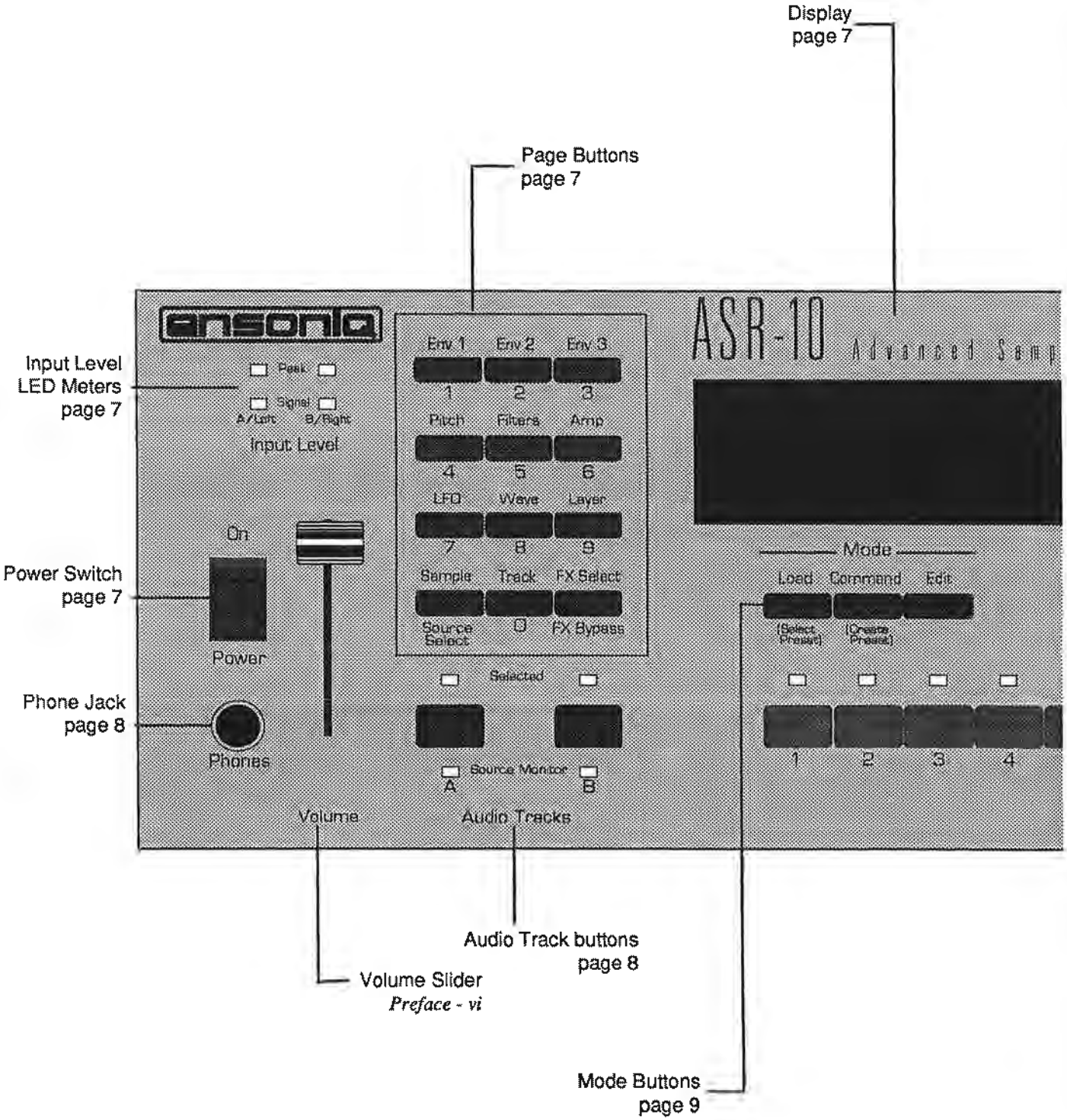
ensoniq®

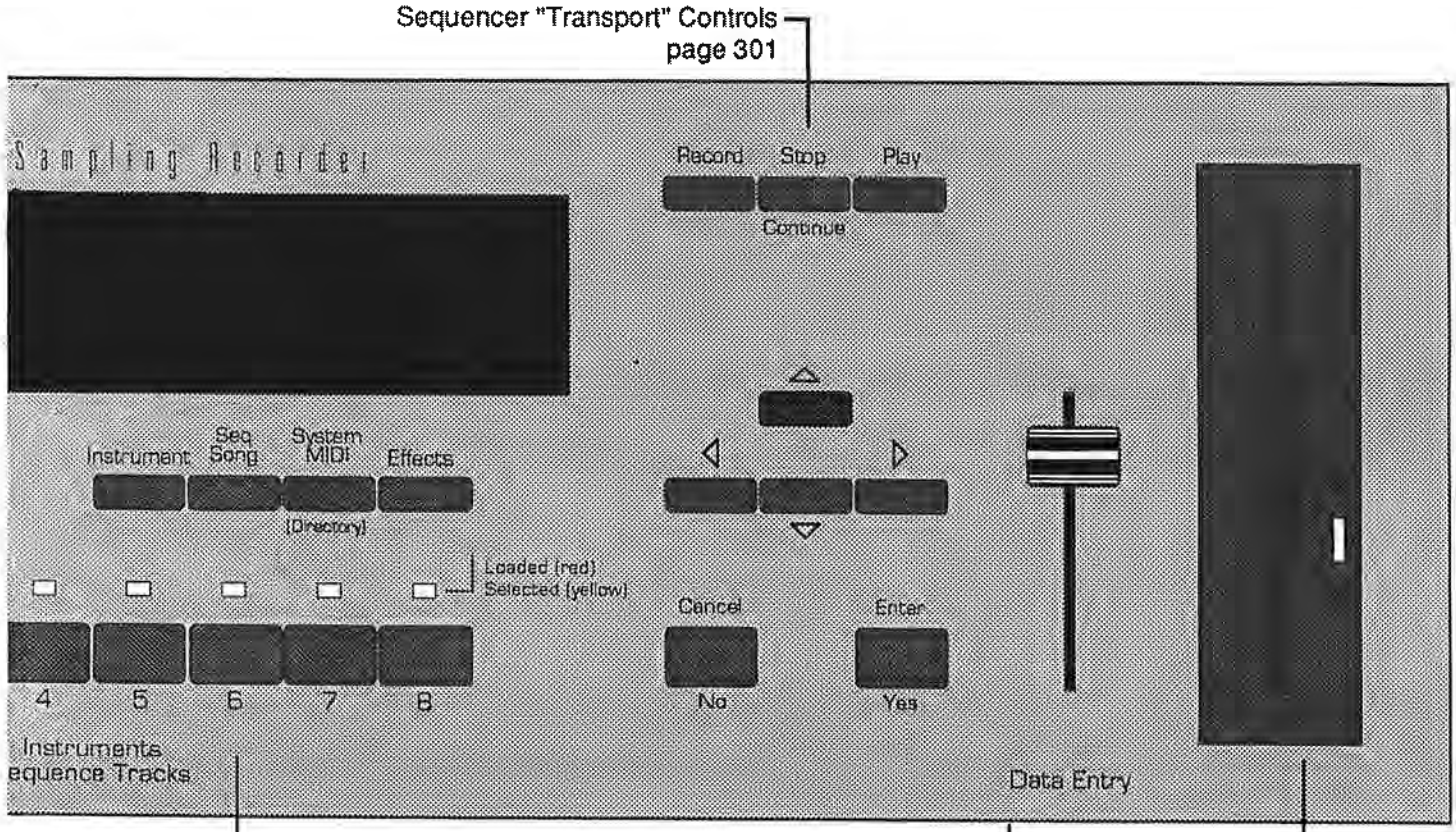
Advanced Sampling Recorder Module

Musician's Manual

Version 1.5

FRONT PANEL CONTROLS:





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ASR–10 Musician's Manual:

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IMPORTANT:

"This equipment generates and uses radio frequency energy and if not installed and used properly, that is, in strict accordance with the manufacturer's instructions, may cause interference to radio and television reception. It has been designed to comply with the limits for a Class B computing device in accordance with the specifications in Subpart J of Part 15 of FCC rules, which are designed to provide reasonable protection against such interference in a residential installation. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures."

* reorient the receiving antenna
* relocate the instrument with respect to the receiver
* move the instrument away from the receiver
* plug the instrument into a different outlet so that the instrument and receiver are on different branch circuits

"If necessary, the user should consult the dealer or an experienced radio/television technician for additional suggestions. The user may find the following booklet prepared by the Federal Communications Commission helpful: 'How to Identify and Resolve Radio-TV Interference Problems.' This booklet is available from the U.S. Government Printing Office, Washington, D.C. 20402 Stock No. 004-000-00345-4."

In order to fulfill warranty requirements the ASR–10 should be serviced only by an Authorized ENSONIQ Repair Station.

The ENSONIQ serial number label must appear on the outside of the Unit or the ENSONIQ warranty is void.

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Part # 9310 0132 01 - A

Model # MM-103

Table of Contents

Table of Contents

Preface

Welcome! i
Sampling i
Synthesis Functions i
Effects i
Sequencing ii
About The Manual ii
Power — Polarization and Grounding iii
Ground Loops iv
AC Line Conditioning iv
MIDI Connections v
Amplification vi
Running Your ASR–10 Through a Home Stereo System vii
Care and Feeding of the Disk Drive vii
Backing-up the O.S. Disk ix
Accessories x
An Important Note About Non-ENSONIQ Accessories and Your ENSONIQ Warranty xi
Need More Help? xii

Section 1 — Controls and Architecture

Rear Panel Connections 3
Front Panel Controls 7
Parametric Programming 11
Architecture 12
"Booting" the ASR–10 12
Memory 13
Disk Memory vs. Internal Memory 13
Memory Expansion 13
What is a SIMM? 14
Internal Memory 14
Expanding the ASR–10 Memory 14
Important Information about SIMMs 15
An Important Note About ElectroStatic Discharge 15
About Instruments 16
Loading an Instrument 16
If You Run Out of System Memory 17
Deleting an Instrument from the Internal Memory 18
About Banks 19
To Load a Bank 20
Playing Instruments 21
Instrument•Sequence Track Buttons 21
Keyboard Range (How Instruments Share the Keyboard in Load Mode) 23
Stacking Instruments 25
Additional Notes 25
Loading Sequencer Data 26
Loading a Song or a Single Sequence from Disk 26
Selecting a Sequence/Song 27
Playing a Sequence/Song 27
Saving Sequencer Data 28

Saving a Single Sequence to Disk 28
Saving a Song (along with all Sequences) to Disk 29
Deleting a File from Disk 29
Managing Your System Headroom 30
Audio Track Headroom 31
Parameter Illustrations 32

Section 2 — System•MIDI

Edit/System•MIDI Page 35
System Parameters 35
FREE SYSTEM BLOCKS 35
FREE DISK BLOCKS 35
MASTER TUNE 35
GLOBAL BEND RANGE 36
ALL NOTES OFF 36
PEDAL (VOLUME / MOD) 36
LEFT FOOT SW 37
AUTO-LOOP FINDING (ON/OFF) 37
MIDI Parameters 38
MIDI BASE CHANNEL 38
TRANSMIT ON (INST CHAN/BASE CHAN) 39
BASECHAN PRESSURE 39
MIDI IN MODE 40
MONO Modes 41
Global Controllers in MONO A and B Modes 41
MIDI CONTROLLERS (ON/OFF) 42
MIDI SYS-EX (ON/OFF) 42
MIDI PROG CHANGE (ON/OFF) 42
To Send a MIDI Program Change from the ASR–10 43
Receiving Program Changes 43
MIDI SONG SELECT (ON/OFF) 43
MIDI XCTRL NUMBER 44
MULTI CONTROLLERS 44
ENTER PLAYS KEY 44
Command/System•MIDI Page 45
FORMAT FLOPPY DISK 45
COPY O.S. TO DISK 46
SAVE GLOBAL PARAMETERS 47
LOAD GLOBAL PARAMETERS 47
CREATE DIRECTORY 48
CHANGE STORAGE DEVICE 48
SAVE MACRO FILE 48
COPY FLOPPY DISK 48
MIDI SYS-EX RECORDER 50
SAVING System Exclusive Data from an External Device 50
LOADING System Exclusive Data from Disk and SENDING to an External Device 51
WRITE DISK LABEL 51
COPY SCSI DRIVE 52
SCSI Drive Hook-Up 52
Using the COPY SCSI DRIVE Command 52
BACKUP/RESTORE 53
BACKUP 54

RESTORE 55
Notes About BACKUP/RESTORE 56
FORMAT SCSI DRIVE 56
Macros 57
What Is a Macro? 57
Loading a Macro File 58
Creating a Macro 58
Invoking a Macro 58
Saving a Macro File 58

Section 3 — Instrument, Bank, and Preset Concepts

Instruments 63
Loading an Instrument 64
Saving an Instrument to Disk 65
Banks 66
Important Notes About Banks 66
To Load a Bank 67
Saving the Contents of Memory as a Bank 68
Performance Presets 69
Selecting Performance Presets 69
Important Notes about Performance Presets 70
Creating a Performance Preset 70
Suggested Uses for Performance Presets 71
Performance Tip — Creating Keyboard Splits 72

Section 4 — Instrument, Bank, and Preset Parameters

Edit/Instrument Page 75
PATCH (Layer Enable/Disable) 75
Using the Patch Select Jack 75
Programming Patches 76
KEYDOWN LAYERS 77
KEYUP LAYERS 77
MIDI OUT CHANNEL 77
MIDI OUT PROGRAM 78
PRESSURE MODE 78
MIDI STATUS 80
SIZE (in blocks) 80
INST NAME 81
PATCH SELECT MODE 81
INST (Key) RANGE 82
XPOS OCT - SEMI (Transpose) 82
Command/Instrument Page 83
CREATE NEW INSTRUMENT 83
COPY INSTRUMENT 83
DELETE INSTRUMENT 83
SAVE INSTRUMENT 84
SAVE BANK 85
CREATE PRESET 86
DELETE INST EFFECT 87

Section 5 — Effect Concepts

Understanding ASR–10 Effects 91
INST (Instrument) Effects 92
BANK Effect 92
ROM Effect 93
Selecting Effects 94
ROM Effects List 94
Sample Rate 96
Instruments and Banks 96
When are New Effects Loaded Into the ESP chip? 96
Loading and Saving ASR–10 Effects 97
Loading an Effect File 97
Saving an Effect to Disk 98
Programming Effects 99
Effect Variations 99
The Effects Busses 99
Effects Mixing 99
Single Function Effect Mixer 100
Dual Function Effect Mixer 100
Triple Function Effect Mixer 101
Performance Control of Tracks When Using Bank Effects 101
Predefined Effect Modulation Parameters 102
Selectable Effect Modulation Parameters 103

Section 6 — Effect Parameters

Command/Effects Page 107
SAVE BANK EFFECT 107
COPY CURRENT EFFECT 107
Edit/Effects Page 108
Effect Variations 108
ROM-01 HALL REVERB 108
ROM-02 44KHZ REVERB 108
ROM-03 ROOM REVERB 108
ROM-04 DUAL DELAYS 110
ROM-05 44KHZ DELAYS 110
ROM-06 CHORUS+REVRB 111
ROM-07 PHASER+REVRB 112
ROM-08 FLANGER+REV 113
ROM-09 ROT. SPKR+REV 114
ROM-10 CHOR+REV+DDL 116
ROM-11 CMP+DIST+REV 117
ROM-12 DIST+CHO+REV 119
ROM-13 WAH+DIST+REV 120
ROM-14 SMALL ROOM 121
ROM-15 LARGE ROOM 121
ROM-16 HALL REVERB2 121
ROM-17 SMALL PLATE 125
ROM-18 LARGE PLATE 125
ROM-19 REVRSE REVRB 127
ROM-20 REVRSE REVB2 129
ROM-21 GATED REVERB 131
ROM-22 NLIN REVRB 1 134

ROM-23 NLIN REVRB 2 134
ROM-24 NLIN REVRB 3 134
ROM-25 MULTITAP DDL 137
ROM-26 EQ + DELAY LFO 139
ROM-27 VCF+DISTORT 142
ROM-28 GUITAR AMP 1 145
ROM-29 GUITAR AMP 2 145
ROM-30 GUITAR AMP 3 148
ROM-31 SPKR CABINET 151
ROM-32 TUNABLE SPKR 152
ROM-33 EQ+CHOR+DDL 153
ROM-34 EQ+VIBR+DDL 156
ROM-35 EQ+FLNGR+DDL 159
ROM-36 EQ+TREM+DDL 162
ROM-37 PHASER+DDL 165
ROM-38 8-VOICE CHOR 167
ROM-39 PITCH SHIFT 169
ROM-40 PITCH+DDL 171
ROM-41 FAST PITCHSH 173
ROM-42 EQ+CMPRESSOR 175
ROM-43 EXPANDER 177
ROM-44 KEYED EXPNDR 179
ROM-45 INVRS EXPNDR 181
ROM-46 DE-ESSER 183
ROM-47 DUCKER 186
ROM-48 RUMBLE FILTR 189
ROM-49 PARAM EQ 190
ROM-50 VAN DER POL 191

Section 7 — Sampling/Signal Source Concepts

What is Sampling? 195
Sample Rate 196
Aliasing 197
Getting Ready to Sample 198
Easy Sampling 198
Using the Audio Tracks During Sampling 200
Setting the Input Level 201
Normalizing Gain 203
Default Output Routings 203
MultiSampling 204
Sampling Over an Existing WaveSample 205
Stereo Sampling 206
Companion Layers 208
Looping 209
Auto-Loop Finding 210
Short (Single-Cycle) Loops 210
Adjusting Loop Position 211
Long Loops 212

Section 8 — Sampling/Signal Source Parameters

Sampling Parameters 217
SAMPLING THRESHOLD LEVEL 217
SAMPLE RATE 218
PRE TRIGGER 219
SAMPLE TIME 219
Additional Multisampling Parameters 220

Section 9 — WaveSample and Layer Concepts

ASR–10 Instrument Configuration 225
About WaveSamples 225
Selecting a WaveSample or Layer for Editing (the Edit Context page)
Editing 227
WaveSample Commands 228
Messages 228
WaveSample Commands in Stereo Layers 228
Modulators 229
About Modulation 229
Selecting a Modulator 229
Modulation Amount 230
Modulation Sources 230

Section 10 — Wave Data Parameters

Edit/Wave Page 237
MODE (WS Playback Mode) 237
SMPL START 238
SAMPLE END 238
LOOPSTART 239
LOOPEND 239
LOOPPOS 240
MOD / SRC 240
MOD AMT / RANGE 241
Command/Wave Page 242
CREATE NEW WAVESAMPLE 242
COPY WAVESAMPLE 242
DELETE WAVESAMPLE 242
WAVESAMPLE INFORMATION 243
TRUNCATE WAVESAMPLE 243
About Loop Commands 244
CROSS FADE LOOP 244
REVERSE CROSS FADE 246
ENSEMBLE CROSS FADE 247
BOWTIE CROSS FADE LOOP 248
BIDIRECTIONAL X-FADE 249
A Note About the COPIES WILL CHANGE Message 250
MAKE LOOP LONGER 251
SYNTHESIZED LOOP 252
CONVERT SAMPLE RATE 253
TIME COMPRESS/EXPAND 254
COPY WAVE PARAMETERS 255
Command/Amp Page (Amplitude Commands) 256

NORMALIZE GAIN 256
VOLUME SMOOTHING 257
About Mix, Merge, and Splice 258
MIX WAVESAMPLES 259
MERGE WAVESAMPLES 259
SPLICE WAVESAMPLES 260
FADE IN 261
FADE OUT 261
Command/LFO Page (Data Commands) 262
CLEAR DATA 262
COPY DATA 262
REPLICATE DATA 263
REVERSE DATA 264
INVERT DATA 264
ADD DATA 265
SCALE DATA 265

Section 11 — WaveSample and Layer Parameters

Edit/Pitch Page 269
ROOT KEY - FINE (Fine Tune) 269
LFO AMOUNT 269
ENV 1 AMOUNT 269
NOISE RATE - AMOUNT 270
BEND RANGE 270
PITCH MOD - AMOUNT 270
WS RANGE LO - HI 270
Command/Pitch Page 271
EDIT PITCH TABLE 271
COPY PITCH TABLE 273
DELETE PITCH TABLE 273
EXTRAPOLATE PITCH TBL 273
ASR–10 Filters 274
Low-Pass and High-Pass 274
Poles — Rolloff Curves 274
Cutoff Frequency 274
Edit/Filters Page 275
MODE (Filter Mode) 275
CUTOFF F1 (Filter 1) - F2 (Filter 2) 277
ENV2 AMT F1 (Filter 1) - F2 (Filter 2) 277
KBD AMOUNT F1 (Filter 1) - F2 (Filter 2) 277
F1 MOD - AMOUNT 277
F2 MOD - AMOUNT 277
Edit/Amp Page 278
WS VOLUME - PAN 278
WaveSample Volume 278
WaveSample Pan 278
VOLUME MOD - AMOUNT 278
PAN MOD - AMOUNT 278
A - B FADE IN - TO 279
C - D FADEOUT - TO 279
Volume Modulator Cross Fade Breakpoints 279
FADECURVE 280
BOOST 280

OUT (Output Routing) 281
ASR–10 Envelopes 282
About ENV 3 282
Typical ASR–10 Envelope 282
Edit/Env 1, Env 2, Env 3 Pages 284
HARD VEL LEVELS 284
SOFT VEL LEVELS 284
TIMES 285
2ND RELEASE (TIME) - LEV 286
ATTACK TIME VEL AMT 286
KBD TIME SCALING 286
SOFT VEL CURVE 287
ENVELOPE MODE 287
ENVELOPE (Envelope Template) 287
Envelope Templates 288
Copying Envelopes 288
Edit/LFO Page 289
WAVE - RATE 289
LFO Wave 289
LFO Rate 290
LFO DEPTH - DELAY 290
LFO Depth 290
LFO Delay 290
LFO MODE 291
DEPTH MOD - AMOUNT 291
RATE MOD - AMOUNT 291
Edit/Layer Page 292
LYR GLIDEMODE 292
LAYER GLIDETIME 293
LEGATO LAYER 293
LYR VEL LO - HI 294
PITCH TBL 294
LYR NAME 294
DELAY - VEL AMOUNT 294
LAYER RESTRIKE TIME 294
STEREO LAYER LINK 295
Command/Layer Page 296
CREATE NEW LAYER 296
COPY LAYER 296
DELETE LAYER 296

Section 12 — Sequencer and Audio Track Concepts

Sequencing on the ASR–10 299
Digital Sequencing 299
What is a Sequence? 300
What is a Song? 300
Banks 300
Sequencer "Transport Controls" 301
Sequencer Status 302
Loading and Saving Sequencer Data 303
Loading a Song or a Single Sequence from Disk 303
Saving a Single Sequence to Disk 304
Saving a Song (along with all Sequences) to Disk 305

Selecting a Sequence/Song 306
Playing a Sequence/Song 306
Sequence Tracks 307
Creating a New Sequence 307
Recording a Sequence 308
"Punching In" on a Track 309
Recording the Default Track MIX Setting 309
Recording Another Track with a Copy of the Same Instrument 310
Additional Notes 311
MIDI Sequencing on the ASR–10 — MIDI Connections 312
MIDI Mode and Channel — Remote MIDI Devices 313
Recording MIDI Tracks 314
Recording Program Changes 314
Recording into the ASR–10 Sequencer from a MIDI Controller 315
Recording Onto a Single Track (One MIDI Channel) 315
Recording Onto Several Tracks (Multiple MIDI Channels) — Multi-Track Record 315
Song Mode 317
Making a Song (Command/Seq•Song, EDIT SONG STEPS) 318
To Go to a Different Step in the Song 320
To Change Anything in an Existing Song Step 320
To Insert a Step Anywhere in the Song 320
To Delete a Step Anywhere in the Song 320
Song Tracks 321
Recording Song Tracks 321
Recording MIXDOWN VOLUME and MIXDOWN PAN 323
About Audio Tracks 324
Selecting, Enabling, and Disabling Audio Tracks 325

Section 13 — Sequencer and Audio Track Parameters

The Edit/Seq•Song Page 329
CURRENT SEQUENCE/SONG 329
Current Sequence/Song 329
BAR/STEP — The GOTO Function 329
TEMPO - LOOP 330
Tap Tempo 330
CLOCK SOURCE 330
CLICK - CLICK NOTE VALUE 331
CLICK VOLUME 331
CLICK PAN - OUT (Output Routing) 331
SEQ COUNTOFF 332
SEQ REC MODE 332
SEQ REC SOURCE 333
Command/Seq•Song Page 334
CREATE NEW SEQUENCE 334
COPY SEQUENCE 334
DELETE SEQUENCE 334
SAVE CURRENT SEQUENCE 335
SAVE SONG AND ALL SEQS 335
RENAME SONG/SEQUENCE 335
SEQUENCER INFORMATION 336
ERASE SONG + ALL SEQS 336
APPEND SEQUENCE 337

CHANGE SEQUENCE LENGTH 337
EDIT SONG STEPS 338
Edit/(sequence) Track Page 339
SEQ TRACK STATUS (Mute/Play/Solo) 339
SNG TRACK STATUS (Mute/Play/Solo) 339
MIX - PAN 340
Track MIX 340
Recording the Default Track MIX Setting 340
Track PAN 341
OUT (Output Routing) 341
EFFECT MOD CONTROL 341
MULTI-IN MIDI CHAN 342
Edit/(audio) Track Page 343
ATRK MIX - PAN 343
Audio Track MIX 343
Audio Track PAN 343
OUT (Audio Track Output Routing) 344
Command/Track Page 345
QUANTIZE TRACK 345
COPY TRACK 346
ERASE/UNDEFINE TRACK 346
FILTER EVENT 347
MERGE TWO TRACKS 348
EVENT EDIT TRACK 348
Event Cursor 349
Current Event Type 349
Key Name 349
Event Value 350
Event Location 350
EVENT TYPE — Event Type Filter 350
Duration Screen for Key Events and Key Pressure Events 351
INSERT and DELETE 351
Auditioning Changes 352
TRANSPOSE TRACK 352
SCALE EVENT 353
SHIFT TRACK BY CLOCKS 353

Section 14 — Storage

Disk Storage — Using the Disk Drive to Save and Load Data 357
ASR–10 Disk File Types 358
Disk Capacity — Bytes, Blocks, and Files 359
ASR–10 Disk Functions 360
FORMAT FLOPPY DISK — Formatting a Blank Disk 360
COPY FLOPPY DISK — Making a Backup Copy of a Disk 361
Loading and Saving ASR–10 Instruments 363
Loading an Instrument 363
If You Run Out of System Memory 364
Saving an Instrument to Disk 364
Loading and Saving ASR–10 Banks 365
To Load a Bank 365
Saving the Contents of Memory as a Bank 366
Loading and Saving Sequencer Data 367
Loading a Song or a Single Sequence from Disk 367

Saving a Single Sequence to Disk 368
Saving a Song (along with all Sequences) to Disk 369
Loading and Saving ASR–10 Effects 370
Loading an Effect File 370
Saving an Effect to Disk 370
A Note About Saving and Loading from a SCSI Storage Device 371
Deleting a File from Disk 371
MIDI System Exclusive Recorder 372
What is System Exclusive Data? 372
System Exclusive Storage 372
Saving Sys-Ex Data Uses the Internal Memory 372
Saving System Exclusive Data from an External Device to Disk 373
System Exclusive Messages 374
Loading System Exclusive Data from Disk to an External Device 375
ASR–10 Disk Messages 376
WARNINGS 376
ERROR MESSAGES 376

Section 15 — Sampling Applications
Resampling through the EQ Effect — Using Equalization 381
Record a Dry Sample 381
Normalizing Gain 383
Setting up the Parametric EQ Effect 383
Resampling with Effect 383
Using the ASR–10 as a Virtual Multi-Track Digital Recorder 385

Section 16 — Instrument Programming Applications
Making a Stereo Sound from a Mono WaveSample 389
Creating Keyboard Cross Fades 391
Velocity Cross-Switching Between Layers 393
Using the ENTER PLAY S KEY Parameter 393

Section 17 — Sequencing/MIDI Applications
Using the ASR–10 as a Master MIDI Controller 397
The MIDI Connection 397
Controlling Remote MIDI devices from the ASR–10 — MIDI Connections 397
Creating a MIDI Instrument 399
Recording into the ASR–10 Sequencer from a MIDI Controller 401
Recording Onto a Single Track (One MIDI Channel) 401
Recording Onto Several Tracks (Multiple MIDI Channels) — Multi-Track Record 401
Using the ASR–10 with an External Sequencer 402
POLY Mode — The ASR–10 as One Instrument 402
MULTI Mode — The ASR–10 as Eight Independent Instruments 403
Some Important Points about MULTI Mode 403
Synchronizing to an External MIDI Clock Source 403
Using the ASR–10 with a Drum Machine 404
To Sync a Drum Machine to the ASR–10 404
To Sync the ASR–10 to a Drum Machine 404

To Sequence a Drum Machine from a Track of the ASR–10 404
Using the ASR–10 with a MIDI Guitar Controller 405
Using MONO A Mode 405
Using MONO B Mode 406
Global Controllers in MONO A and B Modes 407
Patch Selects and Presets 408
Song Position Pointers 409
MIDI Song Selects 409
Delay Times/Tempo BPM Chart 410
The Smart Transmit™ Function 411
Pressure 413

Section 18 — Audio Track Applications
Using a Microphone with the ASR–10 417
Using a Guitar with a Sequence or Song 418
Assigning Different Output Routings 419

Appendix
ASR–10 MIDI Implementation I
Registered Parameters I
MIDI Implementation Chart II

List of Hints

ASR–10 Index

Preface

Welcome!

Congratulations and thank-you for your purchase of the ENSONIQ ASR–10 Advanced Sampling Recorder, another milestone in digital sampling keyboards. The ASR–10 revolutionizes the way sampling is integrated into a workstation, by including 24-bit dynamic effects processing and allowing the stereo audio input to be monitored, sampled, and resampled through the effects. And only ENSONIQ offers all the expressive control that turns a sampler into a truly musical instrument.

Sampling

The ASR–10 features true stereo Sigma-Delta (one-bit) 64 times oversampling technology with a selectable sample rate of 30kHz or 44.1 kHz. Samples can be recorded from a variety of input sources including: the stereo audio inputs, the optional DI-10 Digital I/O Board (consumer AES/EBU digital interface) or the ASR–10's own Main Audio Output (for resampling). The ASR–10 can sample while the sequencer is playing, and can even sample its own sequencer playback. CD quality output circuitry provides state of the art playback performance. The ASR–10 has all the on-board editing functions you'll ever need. Auto looping, volume smoothing, gain normalization, and multiple types of crossfading provide complete control over your sampled sounds.

An advanced Time Compression/Expansion function allows you to change the length or tempo of a sample without affecting the pitch, a necessity for dance music or broadcast needs.

Synthesis Functions

Not only does the ASR–10 provide superb stereo sampling, it is also a full function synthesizer, allowing you to reshape your sampled sounds with maximum flexibility. 31 voice polyphony, multiple filters and envelopes, and a comprehensive modulation matrix gives the ASR–10 the power to manipulate your WaveSamples into many exciting new sounds.

Effects

The ASR–10 incorporates effects processing into a sampler in new and exciting ways — with 50 different 24-bit effect algorithms, the ASR–10 provides a complete arsenal of signal processing and conditioning tools for your use. State-of-the-art Reverbs, Chorusing, Flanging, Phasing, Distortion, Digital Delays, and Speaker Effects, will make your sampled sounds and sequenced music sound totally professional. Signal conditioning effects, like EQ, Compression, and Ducking can be used to process sounds as they are being sampled, to get the perfect sound every time.

The ASR–10 offers continuous audio monitoring of the stereo audio inputs on two dedicated Audio Tracks. The audio inputs can be monitored through the effects and if desired, audio input can be sampled through the effects. Audio Tracks also allow live external signals to play through the effects processor: you can use the ASR–10's high quality effects on other instruments in your rig, or allow fellow musicians to plug right in and play — no mixer required! The ASR–10 also has the ability to resample sounds with effects in real time. For example: add reverb to a drum sample, resample it, then reuse the effects processor to further color your music.

Sequencing

The ASR–10 is more than an expressive sampling instrument: it's also a 16-track sequencing workstation with complete mixdown capabilities. You can record and edit your performances with a wealth of creative tools. And only ENSONIQ sequencers allow you to audition your changes to decide which version to keep. This flexible combination of high fidelity sampling and synthesis, versatile effects processing, and sequencing is what makes the ASR–10 unique.

About The Manual

The Musician's Manual is your guide to unlocking the full power of the ASR–10. At this point, you're probably anxious to plug in your ASR–10 in and get playing. The rest of this section contains important information about power, grounding, amplification and care of the floppy drive. We recommend that you read through this section, and then follow the Tutorial (contained in the seperate Tutorial Manual). The Tutorial is designed to provide step-by-step procedures for performing basic operations on the ASR–10.

After the initial "I just gotta hear it" phase has passed and you're ready to utilize the full potential of the ASR–10, please take the time to read through the sections in this manual on sampling, programming, sequencing, effects, and storage. The Musician's Manual is designed to be used as a reference tool. Both manuals provide valuable information and tips that will speed up the learning process and ensure your maximum enjoyment of this powerful musical instrument.

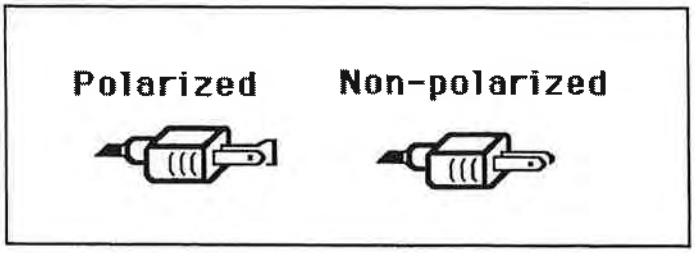
Thank-you again for choosing ENSONIQ. Enjoy the music!

Power — Polarization and Grounding

Like many modern electrical devices, your ASR–10 has a three-prong power cord with earth ground to ensure safe operation. Some products have power cords with only two prongs and no earth ground. To ensure safe operation, modern products with two-prong power cords have polarized plugs that can only be inserted into an outlet the proper way. Some products, such as older guitar amplifiers, do not have polarized plugs and can be connected to an outlet incorrectly. This may result in dangerous high voltages on the audio connections that could cause you physical harm or damage any properly grounded equipment to which they are connected, such as your ENSONIQ product.

To avoid shock hazards or equipment damage, we recommend the following precautions:

- If you own equipment with two pronged power cords, check to see if they are polarized or non-polarized. You might consider having an Authorized Repair Station change any non-polarized plugs on your equipment to polarized plugs to avoid future problems.



- Exercise caution when using extension cords or plug adapters. Proper polarization should always be maintained from the outlet to the plug. The use of polarized extension cords and adapters is the easiest way to maintain proper polarity.

- Whenever possible, connect all products with grounded power cords to the same outlet ground. This will ensure a common ground level to prevent equipment damage and minimize hum in the audio output.

AC outlet testers are available from many electronic supply and hardware stores. These can be used to check for proper polarity of outlets and cords.

Ground Loops

Sometimes currents flowing through the ground line generate a signal seen by another part of the circuit sharing the same ground. In other words, if there are two identical signal paths within a circuit, they can form a loop which can result in hum and/or noise. If you are using equipment that has 3-prong "grounded" AC power cords, you may suffer from a ground loop resulting from the interconnection of this equipment. The following diagram shows how cascading or "chaining" the output of one 3-prong grounded system into the input of another 3-prong grounded system with a standard unbalanced 2 conductor cord (like a 1/4" guitar cable) can result in a ground loop.

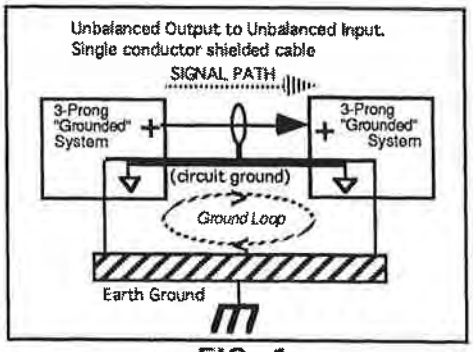


FIG. 1

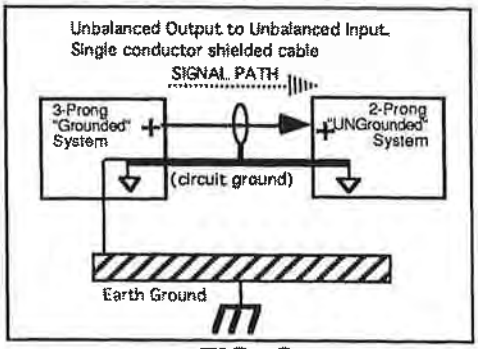


FIG. 2

Fig. 1 depicts a system interconnection where a ground loop can exist. Fig. 2 depicts a system interconnection where a ground loop does NOT exist. When interconnecting 3-prong grounded systems, you can use signal isolation transformers to prevent ground loops. This coupling transformer effectively isolates two interconnected system signal grounds, while still allowing the signal to pass through.

AC Line Conditioning

As is the case with any computer device, the ASR–10 is sensitive to sharp peaks and drops in the AC line voltage. Lightning strikes, power drops or sudden and erratic surges in the AC line voltage can scramble the internal memory and, in some cases, damage the unit's hardware. Here are a few suggestions to help guard against such occurrences:

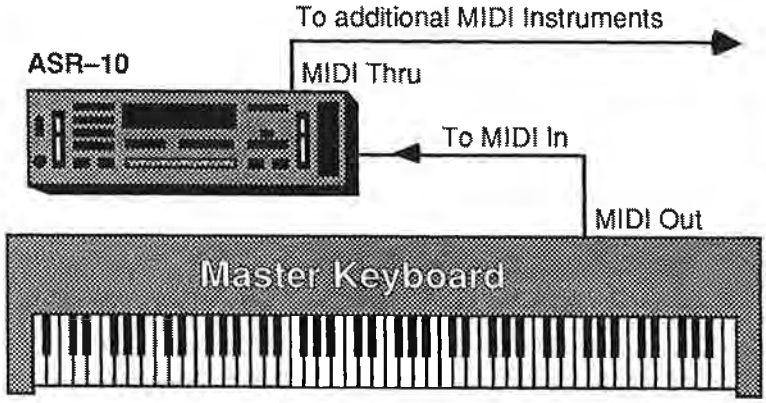
- A Surge/Spike Suppressor. The cheaper of the options, a surge/spike suppressor absorbs surges and protects your gear from all but the most severe over-voltage conditions. You can get multi-outlet power strips with built-in surge/spike suppressors for little more than the cost of unprotected power strips, so using one is a good investment for all your electronic equipment.

- A Line Conditioner. This is the best, but by far the more expensive, way to protect your gear. In addition to protecting against surges and spikes, a line conditioner guards the equipment against excessively high or low line voltages. If you use the ASR–10 in lots of different locations with varying or unknown AC line conditions, you might consider investing in a line conditioner.

MIDI Connections

Of course, you'll need to connect the ASR–10 to a MIDI keyboard or other MIDI controller before it will make any sound at all. Any instrument that has a MIDI Out jack can be used to drive the ASR–10.

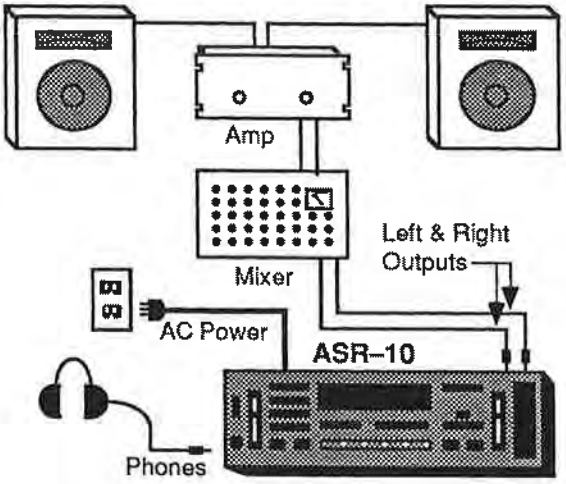
Connect the MIDI Out jack of the master controller to the MIDI In jack of the ASR–10 as shown below, using a standard 5-pin MIDI cable.



If you want to drive additional instruments from the same master keyboard, connect the MIDI Thru jack of the ASR–10 to the MIDI In jack of the next instrument in line. The MIDI Thru jack simply "passes on" the signal received at the MIDI In jack without transmitting any data from the ASR–10 itself.

Amplification

Connect the Main Audio Outputs of the ASR–10 to the line level inputs of a mixer, instrument amplifier, stereo, or any other sound system, using 1/4 inch audio cables. If your system is stereo, connect the Left and Right Main Outputs to two seperate channels of your mixer, stereo, etc. If it's mono, use either of the Main Audio Outputs, but make sure nothing is plugged into the other output. For listening through headphones, plug the phones into the front panel jack marked *Phones*. If you're running the ASR–10 through a mixer in stereo, be sure to pan the left input channel of the mixer fully left, and the right input channel fully right.



It is a good idea to make sure your audio system is turned off (or down) when making connections, to avoid damaging speakers or other components.

Note: The ASR–10 outputs are line-level, and are intended to be connected only to line-level inputs, such as those on a mixer, stereo pre-amp, keyboard amp, etc. Connecting the ASR–10 audio outputs to a mic-level input, such as a guitar amp or the microphone jacks on a tape deck, is not recommended, and might result in damage to the device input.

Move the ***Volume Slider*** *all the way up*. As with any digital musical instrument, the ASR–10 will give the best results if you keep the ***Volume Slider*** full on, and use the volume control on your mixer or amp to adjust its level.

Switch the audio system on, and adjust the amplifier volume for normal listening levels. If you hear no sound while playing the MIDI controller, switch the audio system off and check your connections.

Running Your ASR–10 Through a Home Stereo System

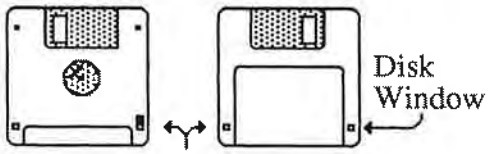
If you are thinking about amplifying your ASR–10 through your home stereo, a word of caution is in order. A home stereo is great for playing CD's, albums, tapes — the dynamic range of these media is limited, and your speakers aren't usually subjected to extreme volume changes and frequency transients. While the dynamic range of CD's is significantly greater than LP's or tapes, the output of a CD player is still conservative compared to the uncompressed, unlimited line-level output of a pro-level keyboard. Running your ASR–10 — or any pro-level keyboard through a home stereo at high volume levels can damage your speakers, not to mention the impedance mis-match this can create. If your only means of amplification is your home stereo, then try to keep your levels on the conservative side.

Care and Feeding of the Disk Drive

The ASR–10's built-in disk drive is used to store all your Instruments, Banks, and Sequencer data, as well as System Exclusive messages from other MIDI devices. The ASR–10 uses a Quad-density disk drive that can store 1600 Kilobytes of data on a Double-Sided High-Density (DSHD) 3.5" micro-floppy disk and 800 Kilobytes of data on a Double-Sided Double-Density (DSDD) 3.5" micro-floppy disk. The disks are enclosed in a protective plastic carrier with an automatic shutter to protect the diskette from physical damage. It is important not to alter this carrier in any way.

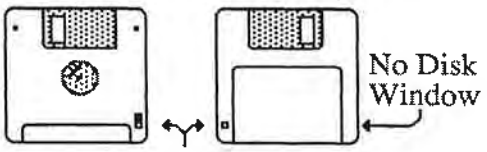
The 3.5" disks have a sliding write-protection tab so that you can protect your sounds and sequences against accidental erasure. Sliding the write-protection tab in the lower left corner of the disk so that the window is closed will allow you to store information on the disk. Sliding the tab so that the window is open will protect the disk against being accidentally reformatted or having files deleted. Double-Sided High Density disks can be easily identified because they have an additional window (with no write-protection tab) located on the lower right corner of the disk.

Double-Sided High-Density (DSHD)



Write Protect Tab

Double-Sided Double-Density (DSDD)



Write Protect Tab

Floppy disks are a magnetic storage medium, and should be treated with the same care you'd give important audio tapes. Just as you would use high quality audio tapes for your important recording needs, we recommend using high quality floppy disks for your ASR–10. Here are a few Do's and Don't's concerning disks and the disk drive.

Do's:

- Use either Double-Sided High-Density (DSHD) or Double-Sided Double-Density (DSDD) 3.5 inch Micro-floppy disks. Both types are available from almost any computer store and many music stores carry them as well.
- Keep your disks and the disk drive clean and free of dust, dirt, liquids, etc.
- Label your disks including the Disk Label name (see *Section 14 — Storage*) and keep a record of what is saved on each.
- Only transport your unit with nothing in the drive.

Don't's:

- Don't use Single-Sided (SSDD or SSSD) disks. These disks have not passed testing on both sides. While a single-sided disk might work successfully with the ASR–10, it is possible that you will eventually lose important data to a disk error if you try using Single-Sided disks.
- Don't put anything other than a disk or the plastic sheet in the disk drive.
- Don't transport the unit with a disk in the drive.
- Don't expose disks to extremes of temperature. Temperatures below 50° F and above 140° F can damage the plastic outer shell.
- Don't expose your disks to moisture.
- Don't dry your disks in a microwave oven.
- Don't subject disks to strong magnetic fields. Exposure to magnetic energy can permanently damage the information on the disk. Keep disks away from speaker cabinets, tape decks, power cables, airline x-ray equipment, power amplifiers, TV sets, and any other sources of magnetic energy.
- Don't eject the disk while the drive is operating (i.e. when the disk drive light is on).

Backing-up the O.S. Disk

Since floppy disks are magnetic media, and are suseptible to magnetic fields, we highly recommend making several back-up copies of your O.S. disk. This can help save time and frustration in the unlikely event that the O.S. Disk becomes damaged. Since the Tutorial files are on the O.S. Disk, we'll use the COPY FLOPPY DISK to back-up all of the information. Here's how:

- Slide open the plastic write-protect tab on the original O.S. disk so that the disk is write protected (you can see through the hole). This is an extra precaution to safeguard the data.
- Press ***Command***, then ***System•MIDI***, and scroll to the COPY FLOPPY DISK command.
- Press ***Enter•Yes***. The display shows INSERT SOURCE DISK (the one you want to copy).
- Insert the source disk, then press ***Enter•Yes***.
- The drive will engage and the display will flash READING SOURCE DISK. Once the drive stops, the display will change to show INSERT DEST DISK.
- Insert the destination disk (the one you want to copy to), and then press ***Enter•Yes***. If the destination disk is unformatted, the display will ask ERASE AND FORMAT DISK? Press ***Enter•Yes*** to format the disk. When formatting is complete, the drive will engage and the display will flash WRITING DEST DISK.
- After writing to the destination disk, the display will read VERIFYING DEST DISK. If the copy is complete, then the display will read DISK COMMAND COMPLETED.

If you do not want to copy all of the Tutorial files, use the Command/System•MIDI COPY O.S. TO DISK command, as described in *Section 2 — System•MIDI*.

We recommend that you use the copied O.S. Disk for daily use, and store the original O.S. Disk in a safe place.

If however, your O.S. disk becomes damaged, and you do not have a back-up copy made, your local Authorized ENSONIQ Dealer will gladly make a new copy for you (you supply the disk).

Accessories

These optional accessories are available from your Authorized ENSONIQ Dealer:

- **ENSONIQ Model SW-10 Dual Foot Switch** — This recommended foot switch can be used for hands-free patch select control (when plugged into the Patch Select jack) or used for effect modulation control, voice sustain, or starting, stopping, and continuing the internal sequencer (when plugged into the Foot Switch jack). Why not buy two (one for each jack) for ultimate control!
- **ENSONIQ Model SW-6 Foot Switch** — A single damper piano style foot switch, for sustain.
- **ENSONIQ Model SW-2 Foot Switch** — A single economical foot switch, for sustain.
- **Model CVP-1 CV PEDAL** — A *Control Voltage Foot Pedal* which can be assigned as a modulator within the voice section of the ASR–10 or used as a volume pedal.
- **AS –Series Sound Libraries** — The AS sound libraries are designed exclusively for the ASR–10, provided on five High Density disks.
- **CDR Series** — These CD ROMs offer a vast array of instruments, banks, and sequence/song arranged in directories by file types. The CDR Series also features Direct Macros, which allows instant access (direct-dial) to any instrument file.
- **SL, SLT, and ESS Sound Libraries** — The ASR–10 can read all of the disks designed for the EPS Series. These disks offer the largest, most accurate, responsive, and musical sampled sounds available anywhere. These sounds are divided into three separate libraries: SL, a series of five-disk sets featuring sounds specifically designed for the EPS–16 PLUS, but fully compatible with the ASR–10; SLT, a series of ten-disk packs originally designed by top sound programmers for the EPS, also compatible with the ASR–10; and ESS, three-disk "Signature Series" sets designed by renowned industry performers and producers like Joey DeFrancesco, Jason Miles, Maurice White, The System, Nile Rodgers, David Hentschel, and others.
- **DI-10 Digital I/O Board (S/PDIF)** — Allowing digital input and output using RCA-type connectors. The DI-10 *must* be installed by an Authorized ENSONIQ Repair Station.

An Important Note About Non-ENSONIQ Accessories and Your ENSONIQ Warranty

ENSONIQ highly recommends that users who wish to add a Digital I/O Interface, or any accessory to their ASR–10 use ENSONIQ-made accessories. However, for those who wish to purchase a non-ENSONIQ product, there are some important things to know about non-ENSONIQ products and your ASR–10's warranty:

- **ENSONIQ will not approve any Non-ENSONIQ Digital I/O Interface for the ASR–10.**
- If your ASR–10 requires servicing, and a non-ENSONIQ accessory is installed, ASR–10 owners will pay a service fee to have it removed so that a technician can diagnose the base unit.
- **If it is determined that the use of an unapproved Non-ENSONIQ product caused the damage, then the repair of that damage is not covered by the ENSONIQ warranty.** Any non-ENSONIQ product which requires opening the case must be installed by an Authorized ENSONIQ Repair Station.
- In addition, if it is found that continued use of a non-ENSONIQ product causes damage to your ASR–10, any future service that your unit might require may not be covered under the ENSONIQ warranty.

Need More Help?

Whether you're an aspiring programmer looking for additional information about basic sampling techniques and MIDI theory, or a professional sound designer working with advanced applications, you may want more detailed information that is beyond the scope of this manual. The following books can help enhance your understanding of sampling, synthesis, MIDI, and related topics. These, in addition to the numerous monthly magazines, provide a wealth of information. While we don't endorse any one of these publications, we offer this partial list as a resource for you to draw on.

The Mix Bookshelf
For prices and more information call: 1-800-233-9604

MIDI
MIDI FOR MUSICIANS, Craig Anderton
MUSIC THROUGH MIDI, Michael Boom
THE MIDI HOME STUDIO, Howard Massey
THE MIDI BOOK, Steve De Furia, Joe Scacciaferro
MIDI RESOURCE, Steve De Furia, Joe Scacciaferro
MIDI IMPLEMENTATION BOOK, Steve De Furia, Joe Scacciaferro
MIDI SYSTEMS & CONTROL, Francis Rumsey
USING MIDI, Helen Casabona, David Frederick
MIDI, THE INS, OUTS AND THRUS, Jeff Rona
MIDI: A COMPREHENSIVE STUDY, Joseph Rothstein

SAMPLING
THE SAMPLING BOOK, Steve De Furia, Joe Scacciaferro

SYNTHESIZERS
GUITAR SYNTH & MIDI, Guitar Player Magazine
SECRETS OF ANALOG AND DIGITAL SYNTHESIS, Steve De Furia
SYNTHESIZER PERFORMANCE & REAL TIME TECHNIQUES, Jeff Pressing
SYNTHESIZER BASICS, Dean Friedman
MUSIC & TECHNOLOGY, H.P. Newquist
A SYNTHESIST'S GUIDE TO ACOUSTIC INSTRUMENTS, Howard Massey

Alexander Publishing
For prices and additional information call: 1-800-633-1123

MIDI
HOW MIDI WORKS, Dan Walker
MURPHY'S LAW MIDI BOOK, Jeff Burger

SAMPLING
EPS–16 PLUS OPERATIONS & SAMPLING, Lorenz Rychner
SAMPLING BASICS, Bobby Maestas

SEQUENCING
SEQUENCING AND ARRANGING Vol's 1-4, Joseph Wagner

SYNTHESIZERS
RECORDING, SYNCING & SYNTHS, Paul Goldfield

Alfred Publishing Company
For prices and more information call 1-818-891-5999
MIDI
ADVANCED MIDI APPLICATIONS, GPI
BASIC MIDI APPLICATIONS, GPI
WHAT IS MIDI?, GPI
SYNTHESIZERS
BEGINNING SYNTHESIZER, GPI
PLAYING SYNTHESIZERS, GPI
SYNTHESIZER PROGRAMMING, GPI

Hal Leonard Publishing
For prices and more information call 1-414-774-3630
MIND OVER MIDI, GPI
SYNTHESIZER TECHNIQUE (REVISED), GPI

Monthly Magazines

The following magazines offer many specific articles and columns that can provide a plethora of useful information.

THE TRANSONIQ HACKER
For prices and more information about this independent news magazine for ENSONIQ Users, call 1-503-227-6848

KEYBOARD
For subscription rates and more information call 1-800-289-9919

ELECTRONIC MUSICIAN
For subscription rates and more information call 1-800-888-5139

HOME & STUDIO RECORDING
For subscription rates and more information call 1-818-407-0744

MIX
For subscription rates and more information call 1-800-888-5139

EQ
For subscription rates and more information call 1-212-213-3444

Controls and Architecture 1
System•MIDI 2
Instrument, Bank, and Preset Concepts 3
Instrument, Bank, and Preset Parameters 4
Effect Concepts 5
Effect Parameters 6
Sampling/Signal Source Concepts 7
Sampling/Signal Source Parameters 8
WaveSample and Layer Concepts 9
Wave Data Parameters 10
WaveSample and Layer Parameters 11
Sequencer and Audio Track Concepts 12
Sequencer and Audio Track Parameters 13
Storage 14
Sampling Applications 15
Instrument Programming Applications 16
Sequencing/MIDI Applications 17
Audio Track Applications 18

Controls and Architecture

System•MIDI

Instrument, Bank, and Preset Concepts

Instrument, Bank, and Preset Parameters

Effect Concepts

Effect Parameters

Sampling/Signal Source Concepts

Sampling/Signal Source Parameters

WaveSample and Layer Concepts

Wave Data Parameters

WaveSample and Layer Parameters

Sequencer and Audio Track Concepts

Sequencer and Audio Track Parameters

Storage

Sampling Applications

Instrument Programming Applications

Sequencing/MIDI Applications

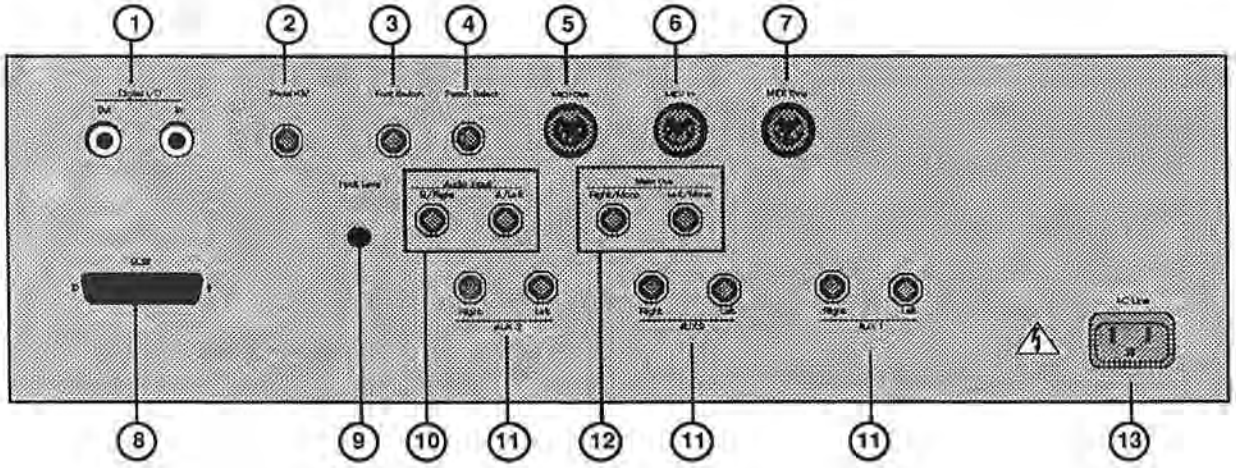
Audio Track Applications

Controls & Architecture

Section 1 — Controls and Architecture

This section provides an introduction to the ASR–10's many controls and rear panel connections, a conceptual overview of the system, a guide to understanding memory, and a discussion of editing various types of parameters. We suggest you read this section carefully — it will help you get the most out of your ASR–10.

Rear Panel Connections



1) Digital I/O — Input/Output

These 2 RCA-type connectors provide direct Digital Input and Output connection to and from the ASR–10 when the optional DI-10 Digital Interface is installed. The Digital Output will provide digital audio output of the Main Output mix — when the current effect uses a 44.1 KHz sample rate — and can also be used for backing-up a SCSI storage device to a DAT (Digital Audio Tape) Recorder equipped with Digital Inputs. The Digital Input can be used for sampling from an external digital audio source and for restoring a DAT backup of an SCSI storage device.

Specs: The Digital Input and Output conforms to the S/PDIF standard.

2) Pedal•CV

This jack is for connecting an optional ENSONIQ Model CVP-1 Control Voltage Foot Pedal, which is assignable as a modulator to various parameters within the ASR–10.

ENSONIQ CVP-1
Control Voltage Foot Pedal

A CV pedal plugged into this jack can also act as a Volume Pedal, controlling the

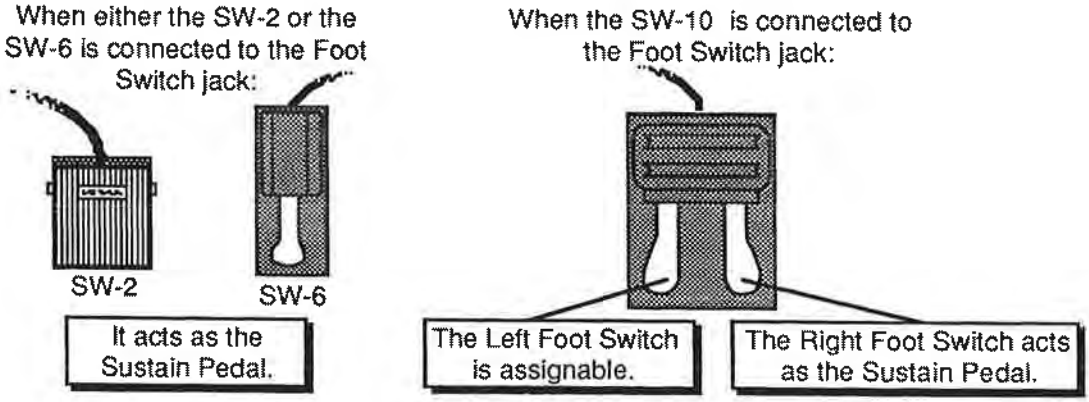
volume of the currently selected ***Instrument•Sequence Track(s)***. A parameter on the Edit/System•MIDI page (press ***Edit***, then ***System•MIDI***, then scroll until the display reads PEDAL=VOLUME MIDI=7), determines whether the CV pedal will act as a modulator or as a volume pedal. Set to PEDAL=VOLUME to use the CV pedal to control volume.

Pedal/CV Specs: 3-conductor (Tip= control voltage input, Ring= 2 KOhm resistor to +12 Volts, Sleeve= ground). 68 KOhm input impedance, DC coupled. Input voltage range= 0 to 10 volts DC. Scan rate= 32mS (maximum recommended modulation input= 15 Hz). For use with an external control voltage, use a 2-conductor cable with the voltage on the tip and the sleeve grounded.

3) Foot Switch

This jack supports either a mono or stereo foot switch, depending on what is plugged into it:

- If you plug either the optional ENSONIQ Model SW-2 Foot Switch or the optional ENSONIQ Model SW-6 Piano Damper Foot Switch into this jack, it will act as a Sustain pedal. Holding it down will cause notes to continue to sustain after the key has been released.
- Or you can connect the optional ENSONIQ Model SW-10 Dual Foot Switch here. The SW-10 is a dual (piano-type) foot switch with two separate pedals. When the SW-10 is connected, the Right Foot Switch will act as a sustain pedal and the Left Foot Switch is assignable.


When either the SW-2 or the SW-6 is connected to the Foot Switch jack:
SW-2
SW-6
It acts as the Sustain Pedal.
When the SW-10 is connected to the Foot Switch jack:
The Left Foot Switch is assignable.
The Right Foot Switch acts as the Sustain Pedal.

A parameter on the Edit/System•MIDI page (press ***Edit***, then ***System•MIDI***, then scroll until the display reads "LEFT FOOT SW=OFF") determines the function of the Left Foot Switch.

Hint: The Sustain pedal can be used to dynamically "latch" the current amount of pressure being exerted on the keyboard. Here's how:

1) Select a sound that responds to pressure. Choose a sustaining sound like an organ.
2) Connect the MIDI Out jack of a pressure-sensitive MIDI keyboard to the MIDI In jack on the ASR–10.
3) Play a key on your controller keyboard and press into the keyboard until you can hear the pressure modulation affect the sound of the note.
4) Press and hold the Sustain pedal (on either your controller keyboard or on a foot switch connected to the ASR–10's Foot Switch jack.
5) Release the key. You will hear that the sound continues to be modulated by pressure at the depth to which you were pressing.
6) Play a different key on your controller keyboard. Notice that the new note is not modulated. You can now press into the keyboard and modulate the new note independent of the note that is sustained. When the current pressure output exceeds the latched level, pressure on the new note will modulate both notes.
7) To release the "latched" pressure value on the sustained note, either press the "latched" key again, or release the Sustain pedal.

4) **Patch Select** (Foot Switch)

By connecting the optional SW-10 Dual Foot Switch to this jack, you can select alternate groups of voices (called *Layers*) within a sound. The ASR–10 can be programmed so that the sound changes (sometimes in subtle ways, sometimes radically) when you play notes with one or both Patch Select foot switches held down. As you play instruments on the ASR–10, make sure you explore what these foot switches do to each sound.

5) **MIDI Out**

This jack transmits MIDI information generated by the ASR–10 sequencer and/or Smart Trasmit™ function to other instruments and computers.

6) **MIDI In**

This jack receives MIDI information from other MIDI instruments or computers.

7) **MIDI Thru**

This jack "passes on" all MIDI (Musical Instrument Digital Interface) information received by the ASR–10 to other MIDI devices. Information generated by the ASR–10 itself does not go to this jack — the Thru jack merely echoes what comes into the MIDI In jack.

8) **SCSI Connector**

The SCSI (Small Computer Serial Interface) connector allows the ASR–10 to transfer data to and from a SCSI-compatible hard disk, CD ROM player, or exchange information with computers at very high speed. Hard drives provide a faster, more convenient way to store instrument, sequencer, bank, and MIDI files.

9) **Input Level Trim Control**

This knob allows you to amplify the level of the external signal source. Use this knob to bring the external signal source up to clipping level.

10) Audio Input — B/Right and A/Left

These jacks are the Right and Left Audio Inputs into the ASR–10 for sampling or Audio Track monitoring of external analog audio sources.

SPECS: 140 KOhm input impedance, AC coupled. The ASR–10 will accommodate line-level signals from +15.5dBV (Input Level Trim control fully counterclockwise) to -16.5dBV (Input Level Trim control fully clockwise).

11) Auxiliary Outputs — Aux 1, Aux 2, Aux 3

These six Auxiliary Outputs provide three pairs of stereo outputs (or 6 individual outs) *in addition* to the main stereo outputs. Each WaveSample, or an entire ***Instrument•Sequence Track***, can be assigned to any of the three Aux stereo pairs and can be independently panned within the stereo field.

12) Main Out — Right/Mono and Left/Mono

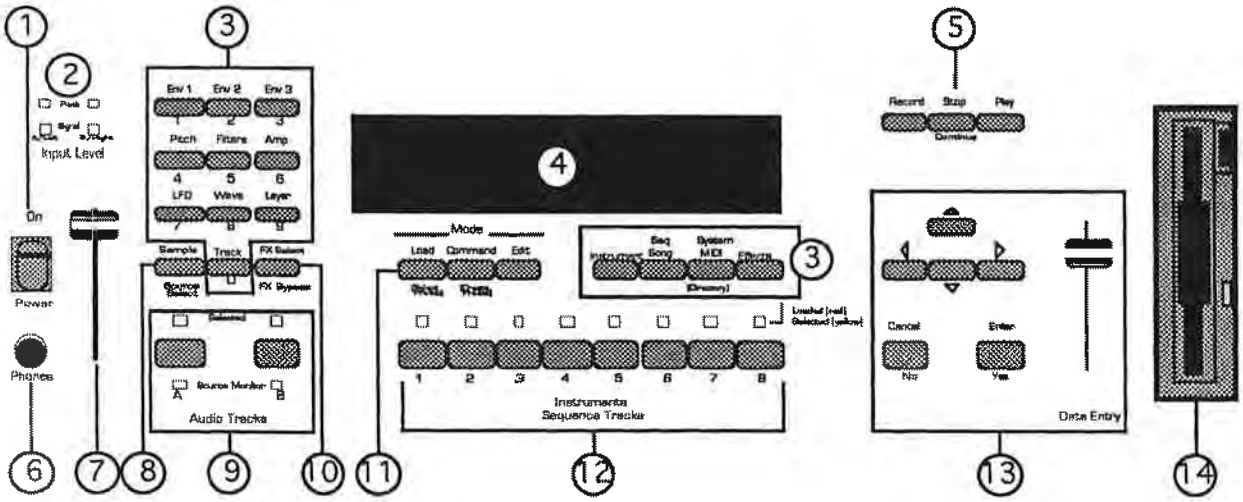
To operate the ASR–10 in stereo, connect these outputs to two seperate channels of your mixer and pan the mixer channels right and left. Note that *either* of the main audio outputs can be used as a mono output. If you want to listen to the output in mono, make sure that only one of the Main Output jacks is connected.

13) AC Line In

The supplied line cord connects here. The correct voltage for the ASR–10 is on the rear panel along with the serial number. If you travel, remember the ASR–10 will only operate on the listed voltage.

Front Panel Controls

Almost everything you do on the ASR–10 — whether it's selecting a sound, editing that sound, adjusting the tuning, etc. — is controlled from the front panel using the following controls:



1) Power

The power switch turns the ASR–10 on and off. When you turn the power on, the display lights up and shows "PLEASE INSERT DISK," which is the prompt to load the operating system.

2) Input Level LED Meters

These 2 dedicated Input Level meters provide separate Left/Right metering of the Audio Input levels, pre-FX, at all times. The green Signal LEDs light at -24 dB. The red Peak LEDs light at 6 dB below clipping.

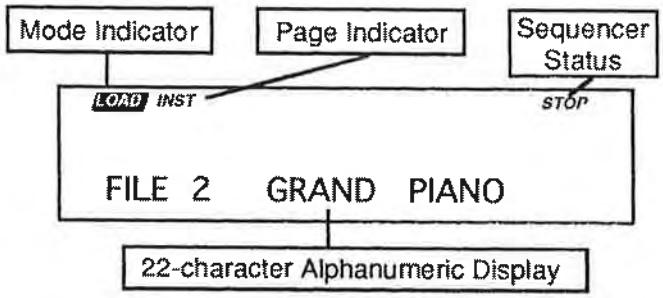
3) Page Buttons

Within each mode, the available disk files, commands, and parameters are organized into *Pages*. A page is selected by pressing one of these fourteen page buttons. Once you are on the correct page, use the ***Data Entry Controls*** to scroll through the files, commands or parameters on the page. A given page will have different functions depending on the current mode. Each mode has a different set of pages available. Not all fourteen page buttons are active in all three modes.

The ten *numbered* page buttons also double as a numeric keypad for "direct-dialing" a given disk file, command, or parameter or for sending MIDI Program Changes.

4) Display

The ASR–10 Display is divided into two main sections: the *Indicator Lights* in the top half of the window and the *22-character Alphanumeric Display* at the bottom of the window.



The indicator lights will tell you which mode the ASR–10 is in (Load, Command, or Edit); which page it is on; and the sequencer status (Stop, Play, Record, etc.). The 22-character alphanumeric display is used to show you information about specific files, commands, parameters, etc. It will also ask you for additional input when necessary, such as which track you want to load an instrument into, or which WaveSample you want to edit.

5) Sequencer "Transport Controls"

These three buttons are used to control the ASR–10's internal multi-track sequencer.

6) Phones

To listen to the ASR–10 in stereo through headphones, plug the phones into this jack. The phones output contains a mix of the signal from the main outputs. Headphone volume is controlled by the volume slider on the front panel. Note that plugging headphones into this jack *does not* automatically turn off the audio in the right and left outputs.

7) Volume Slider

This controls the overall volume of the ASR–10 audio outputs.

8) Sample•Source Select Button

This button is used to initiate sampling (digitally recording sounds) by the ASR–10. The Record Source selected on this page will determine the audio signal that will be monitored on the Audio Tracks.

9) Audio Track Buttons — A and B

The two ***Audio Track*** buttons allow continuous audio monitoring of the stereo audio inputs through effects (if desired), enabling you to sing along (or play a guitar) through effects while playing a sequence, for example. Each Audio Track has its own Edit/Track MIX, PAN, and OUT bus assignment. The signal monitored on Audio Tracks is the signal that will be sampled.

Each ***Audio Track*** button contains 2 LEDs:

- The lower LED is red, and is labelled "Source Monitor." When lit, it indicates that the Audio Track Record Source (set on the Sample•Source Select page, REC SRC parameter) can be monitored on the Audio Track, and that a voice is being used to monitor this audio signal. When the Source Monitor LED is off, the Audio Track is muted and inaudible.

When REC SRC= MAIN-OUT, the audio inputs are disabled, and both Source Monitor LEDs remain off at all times. This happens because REC SRC= MAIN-OUT recording is all ASR–10 generated audio that is routed to BUS1, 2, or 3, and the Source Monitor voices are not needed to monitor this REC SRC, as it is always audible out the Main Outs.

- The upper LED is yellow, and is labelled "Selected." When lit, it indicates that the track is selected for parameter editing, and that the Audio Signal Source being monitored on the track is selected for sampling. When the Selected LED is off, the track's Edit/(audio) Track parameters cannot be edited, and the Audio Signal Source being monitored on the track will not be sampled.

See *Section 12 — Sequencer and Audio Track Concepts* for more information on Audio Tracks.

10) FX Select•FX Bypass Button

This button acts as the "master" control switch for the built-in effects, determining which, if any effect will be used and how it will interact with the instruments that reside in the internal memory. The controls on this page also determine the current sample rate and polyphony.

11) Mode Buttons

These three buttons are the key to finding your way around the ASR–10. The ASR–10 is *always* in one of these three *Modes* — LOAD, COMMAND, or EDIT. The current mode is selected by pressing the appropriate mode button. The highlighted word in the upper left corner of the display tells you which is the current mode.

- LOAD mode is the one you will be in most often — since the ASR–10 lets you continue playing while loading sounds and sequences, LOAD mode also doubles as the "Performance" mode. When the LOAD indicator is flashing, the display is showing you disk files for loading. When the LOAD indicator is lit but not flashing, the display is showing you the name(s) of the instruments in the Internal Memory.

Hint: In flashing LOAD mode (when the LOAD indicator is flashing), successive presses of the ***Instrument***, ***Seq•Song***, ***System•MIDI***, or ***Effects*** buttons will scroll through the available files of that type on the selected storage device.

- COMMAND mode is used to execute a wide variety of commands, such as: saving instruments, banks, and sequences to disk; copying instruments, layers, and WaveSamples from one internal location to another; creating and modifying sequences and songs; and manipulating WaveSamples and their loops in various ways... just to name a few.

Hint: In COMMAND mode, successive presses of each of the 14 page buttons will scroll through the Command screens, one at a time.

- EDIT mode is used to select and modify a great many variables — or parameters — ranging from the volume of a WaveSample, to the velocity response of the instrument, to the MIDI In Mode. Edit mode is also the mode in which all sequence recording and mixing is done.

Hint: In EDIT mode, successive presses of each of the 14 page buttons will scroll through the Edit screens, one at a time.

12) Instrument•Sequence Track Buttons — 1 through 8

These eight buttons are used to select, deselect, and "stack" the various instruments that are loaded into the internal memory of the ASR–10. For each of the eight locations, the LED above the button indicates whether an instrument is *Loaded* into that location (LED shows red) and whether it is *Selected* (LED shows yellow). See "*Playing Instruments*" later in this section for a full discussion of the ***Instrument•Sequence Track*** buttons.

Each instrument location is also a sequencer track — that is, whatever is recorded on track 1 of a sequence will play the instrument that is loaded into location #1. When you are recording, editing or mixing sequences and songs, you use these buttons to select the current track.

13) Data Entry Controls

Once you are in the desired mode and have selected the proper page, you use the controls in the data entry section to: locate and load the desired file (in Load mode); locate and execute the desired command (in Command mode); or locate and modify the value of the desired parameter (in Edit mode).

- The ***Data Entry Slider*** and the ***Up*** and ***Down Arrow*** buttons will: move through the files on the current disk or directory (in flashing LOAD mode); change the value of the current parameter (in Edit mode); or respond when the ASR–10 asks you for further input during the execution of a command (in Command mode).
- The ***Left*** and ***Right Arrow*** buttons are used primarily to move to the next parameter or command on the current page.

Hint: To advance by screens (instead of by parameters), while holding down the ***Right Arrow*** button, press the ***Up Arrow*** button, or while holding down the ***Left Arrow*** button, press the ***Down Arrow*** button.

- The ***Enter•Yes*** and ***Cancel•No*** buttons are used to either proceed with or cancel the function currently showing on the display. The ***Enter•Yes*** button is also used to select a default ROOT KEY when sampling, and can be used to play a note (as set by the Edit/System•MIDI, ENTER PLAYS KEY parameter). Think of the ***Enter•Yes*** button as a "wee-little keyboard."

Hint: When editing any parameter, pressing ***Cancel•No*** will reset the parameter to the value it was set to before it was last edited.

14) Disk Drive

The disk drive is used to store all your Instruments, Banks, and Sequencer/Song data, as well as System Exclusive messages from other MIDI Devices. The ASR–10 disk drive accepts both Double-Sided Double-Density (DSDD) disks as well as Double-Sided High-Density (DSHD) disks.

Parametric Programming

The method used to modify or edit programs, presets and system parameters is called *Page-driven Parametric Programming*, which sounds like a mouthful, but don't worry. Once you've grasped a few basic concepts you'll find that operating the ASR–10 is quite simple, given its many capabilities.

It is likely that you have already encountered some form of parametric programming on other synthesizers or samplers. What this means is that instead of having a separate knob or slider for each function, you have one master ***Data Entry Slider*** and the ***Up/Down*** and ***Left/Right Arrow*** buttons, that adjust the value of whichever parameter you select.

This approach has many advantages, the most obvious is that it greatly reduces the amount of hardware — knobs, switches, faders, etc. needed to control a wide variety of functions. If the ASR–10 had a separate control for each function, it would literally have hundreds of knobs.

Architecture

"Booting" the ASR–10

Insert the power cord into the line receptacle on the back of the ASR–10, next to the power switch. Plug the other end of the cable into a grounded AC outlet. The proper voltage for your ASR–10 is listed on the Serial Number label on the rear panel. Turn the ASR–10 power on and make sure the display lights up. If not, check your connections and power source.

The ASR–10 Operating System (O.S.) — the computer program that tells the hardware what to do — is "disk based." This means that each time you turn the ASR–10 on, the first disk you insert must be one containing an ASR–10 Operating System (see the disk label). In computer-speak, this is called "booting" the machine. Insert the disk with the label facing up and the sliding metal door facing away from you. The display will read LOADING SYSTEM while the O.S. is being loaded. You should always use the latest (highest-numbered) Operating System. If the first disk you put in the drive doesn't contain the ASR–10 Operating System — the display will flash O.S. NOT ON DISK or DISK NOT FORMATTED. Just remove that disk and insert a proper ASR–10 O.S. disk.

Note: We recommend that you use a *copy* of the original O.S. Disk for daily use, and store the original O.S. Disk in a safe place. For more information, see the *Preface*.

Once it has "booted" the ASR–10 is ready to operate, but it won't make any sound until you LOAD an instrument into its internal memory and then select that instrument by pressing its ***Instrument•Sequence Track*** button. These functions will be covered later in this section.

Memory

Disk Memory vs. Internal Memory

The instruments, banks, and sequences that the ASR–10 plays are stored on 3.5" micro-floppy disks. The ASR–10 uses a high-density (HD) drive, allowing you to uses both Double-Sided High-Density and Double-Sided Double-Density disks:

Disk Type	High-Density	High-Density	Double-Density	Double-Density
Format (Sector Offset)	ENSONIQ (offset 0)	COMPUTER (offset 1)	ENSONIQ (offset 0)	COMPUTER (offset 1)
Kilo Bytes	1600	1440	800	720
Sample Words	800k	720k	400k	360k
Blocks	3176	2863	1585	1426

A Block is a handy unit that the ASR–10 uses to measure Internal and Disk memory — 1 Block=256 sample words; 4 Blocks=1k sample words.

Sounds and sequences must be loaded from the disk into the internal memory of the ASR–10 before they can be played. Once it's loaded into memory, an ASR–10 sound or sequence is completely independent of the copy on the disk — you can do anything you want to it without harming the version on the disk, unless you intentionally save the changes. You should feel free to experiment as much as you like with the instrument, layer, and WaveSample parameters of any sound that came with the ASR–10. As long as you have it safely on the disk, you can just reload it and start over if your experiments go awry.

Important: The data in the ASR–10 internal RAM Memory is ***not*** retained when the power is turned off. Anything in memory, whether instruments, banks, or sequencer data, *must* be saved to disk before you switch the power off, or it will be gone forever.

Memory Expansion

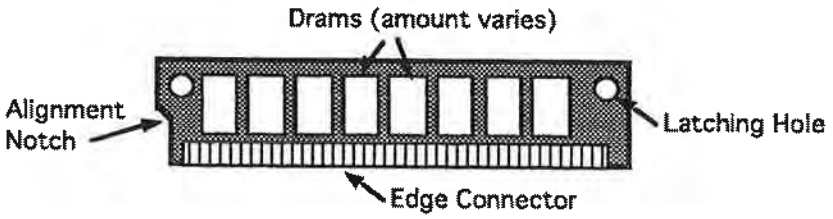
If you want to expand the memory, the ASR–10 can address up to 16 MegaBytes/8 MegaWords using industry standard 1m x 8 or 4m x 8 non-parity SIMMs.

Warning!

Because expanding the memory requires opening the ASR–10 casing (which voids the ENSONIQ Warranty), memory expansion is not user-installable, and must be installed by an Authorized ENSONIQ Repair Station. To prevent damaging your ASR–10, and/or causing bodily injury, do not attempt to install the SIMMs yourself.

What is a SIMM?

SIMM is an acronym which stands for Single In-line Memory Module. SIMMs have become the industry standard used by most computers (both IBM and Mac compatible) to expand the computer's memory. Because of this, SIMMs are readily available in most computer software stores, and from mail order companies. The ASR–10 memory, like a computer, is also expanded using SIMMs.



Internal Memory

As it comes out of the box, the ASR–10 contains 2 MegaBytes or 1 MegaWord of internal memory (a *word* is one single sample, or 16 bits). That's enough for 31.5 (mono) or 15.75 (stereo) seconds of sampling at a 29.8 KHz sample rate, or about 400,000 notes of sequencer memory.

This internal memory is shared by sounds and the sequencer. The memory is distributed *dynamically* between instruments and sequences, which means that the more sounds you have in memory, the less sequencer memory you have, and vice versa.

Expanding the ASR–10 Memory

If you want to expand the memory, the ASR–10 can address up to 16 MegaBytes/8 MegaWords, using industry standard 1m x 8 or 4m x 8 non-parity SIMMs. There are five different memory allocations, as shown below:

SIMMS	1m x 8 (standard)	4m x 8	1m x 8	1m x 8 & 4m x 8	4m x 8
# of SIMMs Used	two	two	four	two & two	four
Mega Bytes	2	8	4	10	16
Mega Words	1	4Z	2	5	8
Blocks	3,800*	16,000*	7,900*	20,000*	31,000*

* Actual block count may vary in different O.S. Versions.

Important Information about SIMMs

- The ASR–10 was designed to use 1m x 8 or 4m x 8 non-parity SIMMs (not 1m x 9 or 4m x 9 parity SIMMs for IBM PC compatibles). We do not recommend using parity SIMMs. These SIMMs may not operate properly, and may cause damage to the ASR–10.
- When expanding the memory, the ASR–10, will only accept D-RAM SIMMs in the expansion slots. The ASR–10 will not accept static RAM or ROMs.
- Any configuration that does not use two or four SIMMs will not work (the system will not even boot up).

An Important Note About ElectroStatic Discharge

SIMMs are susceptible to ElectroStatic Discharge (ESD) commonly known as "static." ElectroStatic Discharge can destroy or damage SIMMs. Since you will buy your SIMMs from a computer store, before you bring them to an Authorized ENSONIQ Repair Station for installation, here are some procedures you can follow when handling SIMMs in order to minimize the possibility of causing ESD damage:

1) Before handling SIMMs, you should be grounded by using a ground strap to discharge any static electric charge built up on your body. The ground strap attaches to your wrist and a ground source allowing your hands to be free to work.
2) Avoid any unnecessary movement, such as scuffing your feet when handling SIMMs, since most movement can generate additional charges of static electricity.
3) Minimize the handling of the SIMMs. Keep them in their static free packages until needed. Only transport or store the SIMMs in their protective packages.
4) When handling the SIMMs, avoid touching the connector pins. Try to handle the SIMMs by the edges only.

If you have any questions concerning the use of SIMMs, the ASR–10, or for additional technical support, please contact ENSONIQ Customer Service at (215) 647-3930 Monday through Friday 9:30 a.m. to 12:15 p.m. and 1:15 p.m. to 6:30 p.m. Eastern Time.

About Instruments

We refer to ASR–10 sounds as *Instruments*. A grand piano, an electric bass, a multi-sampled drum set, a complete string section — each of these would be an example of an instrument. You can load up to eight instruments into the ASR–10, memory permitting, and have instant access to any or all of them.

Each instrument contains four different *Patches* that are selected when the optional SW-10 Dual Foot Switch is connected to the Patch Select jack. These patches allow a single instrument to have four different inflections, voicings, tunings, or synth-type program variations all available with the press of a foot switch.

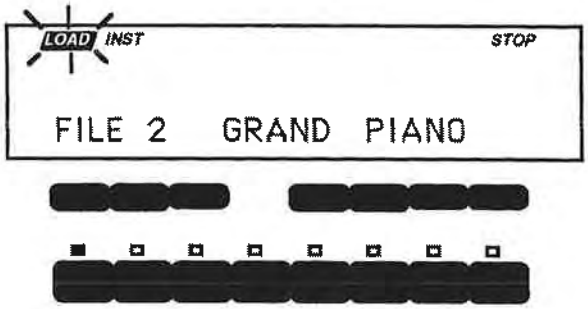
An instrument can be any size (within the limits of memory) — one instrument might consist of a single WaveSample that plays over the entire keyboard, while another might have as many 127 different WaveSamples.

For controlling remote devices, you can create an instrument that contains no samples at all and assign it to play *only* out MIDI, on a particular MIDI Channel.

Loading an Instrument

You can load up to eight different instruments into the ASR–10 at once (within the limits of memory). First, insert a disk containing one or more instrument files into the disk drive.

- Press ***Load.*** The LOAD indicator flashes.
- Press ***Instrument.*** The display looks like this:

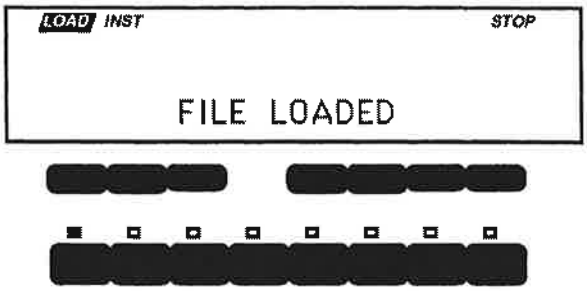


When the LOAD indicator is flashing, the ASR–10 is showing you disk files (think of it as a question mark — the ASR–10 is saying "Load the file showing on the display?"). Pressing the ***Up/Down Arrow*** buttons takes you through the files on the disk. If there are none, the display will read "NO INST OR BANK FILES."

Whenever a disk file is displayed as above, you can press the ***Left*** or ***Right Arrow*** button to see the size of that file in *Blocks* (a Block is 256 samples; 4 Blocks=1K sample words). Press the ***Left*** or ***Right Arrow*** button again to return to the file name.

- Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to view the various instrument files on the disk. Each file has its own File Number. When an instrument file is showing, the INST indicator is lit. The BANK Indicator will light when a bank file is showing.

- Find the instrument you want to load, and press ***Enter•Yes***. The display will say PICK INSTRUMENT BUTTON. The ASR–10 is asking in which instrument location you want to load the sound into.
- Press any of the eight ***Instrument•Sequence Track*** buttons. The ASR–10 will begin immediately loading the instrument into the selected location. The display reads LOADING FILE... and the LED *flashes* red while the instrument is being loaded.



Once the instrument has been loaded, the display briefly shows "FILE LOADED." The LED above the ***Instrument•Sequence Track*** button stops flashing and remains solidly lit red, indicating that there is now an instrument loaded in that location which can be selected by pressing that button.

If you tell the ASR–10 to load an instrument into a location that already has an instrument loaded (LED red), the new instrument will be loaded into that location and the one that was there will be automatically deleted.

If You Run Out of System Memory

You might have to delete an instrument(s) before loading the new one. If there are already one or more instruments loaded into the ASR–10, there might not be enough free memory to load the new one. In this case, the display will say PICK INST TO DELETE. At this point you have three choices. You can:

1. Press any loaded ***Instrument•Sequence Track*** button. That instrument will be deleted from memory and the new one will be loaded; or,
2. Press ***Enter•Yes***. The ASR–10 will delete an instrument (or instruments, as needed) for you, starting from the *highest-numbered* one in memory; or,
3. Press ***Cancel•No***. The Load command will be canceled with no harm done.

Note: You can continue to select and play existing instruments while the new one loads. No more "down-time" waiting for the next sound to load. With the ASR–10, you can be loading the next sound you need while continuing to play the currently selected sound.

Deleting an Instrument from the Internal Memory

Sometimes you will want to delete an instrument from memory — to free up some memory for sampling, for instance. Make sure you have saved the instrument to disk before deleting it. Here's how:

- Verify that the instrument you want to delete is selected.
- Press ***Command***, then ***Instrument***.
- Press the ***Right Arrow*** button repeatedly until the display reads DELETE INSTRUMENT.
- Press ***Enter•Yes***. The display asks DELETE <INST NAME>?
- Press ***Enter•Yes*** (or press ***Cancel•No*** to abort the procedure).

Hint: There is a "shortcut" method for deleting an instrument from internal memory. First, press the ***Instrument•Sequence Track*** button of the instrument you would like to delete. Then, *while holding down the* ***Instrument•Sequence Track*** *button*, press ***Cancel•No***.

About Banks

Banks provide a way to load a whole group of instruments and sequences into the ASR–10 with a few button presses. When you save a bank to disk, it is like taking a "snapshot" of the contents of the ASR–10 Internal memory — the bank file contains information about which instruments are loaded into which of the eight ***Instrument•Sequence Track*** locations, and which song and sequences (if any) are currently loaded in Internal Memory. When you load a bank, the ASR–10 will set up Internal Memory in the same state it was in when you saved the bank file, loading those instruments into the same locations and, if you choose, loading the song with its related sequences. A bank will also save any Performance Presets you have created.

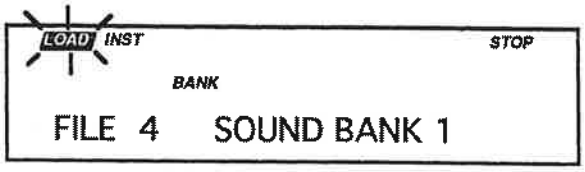
For example, you might have a piano loaded into ***Instrument•Sequence Track 1***, a bass in ***Instrument•Sequence Track 2***, and drums in ***Instrument•Sequence Track 3***, and you have created a number of Performance Presets containing different keyboard configurations of those instruments. Let's say you also have in memory a song, which is composed of 12 sequences. If you now save the contents of memory as a bank, you can later call up this exact setup by loading the bank.

The bank file on a disk doesn't contain the actual instrument and sequence files — it is just a set of instructions telling the ASR–10 what files to load and where to load them. All the instruments and the song in a bank must be saved to disk individually before saving the bank.

Note: Any instruments already loaded into locations not used by the bank will be left intact (memory permitting). You can select and play such instruments while the bank loads.

To Load a Bank

- Press ***Load***, then ***Instrument***.
- Use the ***Up/Down Arrow*** buttons to view the different files.



When a bank file is showing, the BANK indicator lights on the display along with the INST indicator.

- Once a bank file is showing, press ***Enter•Yes***. The ASR–10 will begin loading the instruments and the song data. As it loads each instrument, the display tells you what it's doing. As soon as any of the instruments are finished loading, you can select that instrument and play while the rest of the bank loads.

At some point in the loading process, you may encounter the following message:

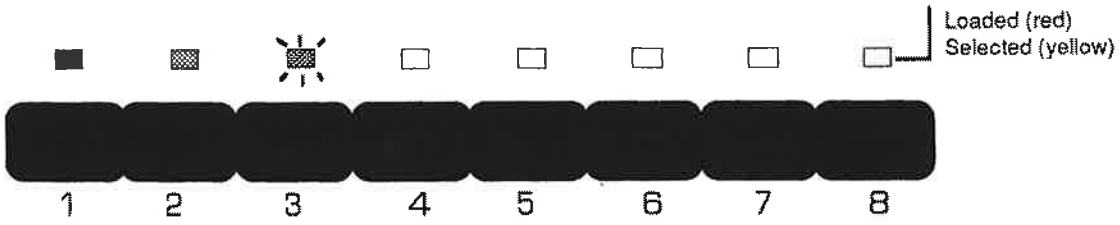
This prompt is informing you that an instrument or song saved as part of the bank is on a different disk than the disk currently in the drive. When this occurs:

- Eject the current disk in the disk drive and replace it with the disk that matches the Disk Label ID requested by the ASR–10.
- Press ***Enter•Yes***. The ASR–10 will resume loading until completed, or until the point when it needs another disk.
- When it has finished loading the instruments, the ASR–10 will load the song (if any) and then set up any copied instruments included in the bank.

Playing Instruments

- First, press ***Load*** until LOAD is solidly lit in the display to make sure you are in LOAD Mode.

Now we'll look at the eight ***Instrument•Sequence Track*** buttons and how they function in LOAD Mode.



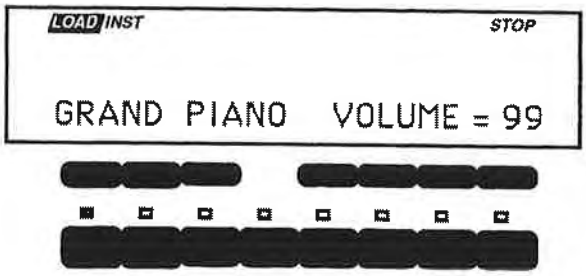
Instrument•Sequence Track Buttons

These buttons represent a location or "slot" into which an instrument can be loaded. The LED above each button tells you if there's an instrument loaded into that location and whether it's selected, deselected or "stacked."

- The LED lights *red* to indicate that an instrument is loaded into that location and can be selected by pressing that button. If the LEDs are not lit, no instruments are loaded. The LED *flashes* red while an instrument is being loaded from disk.
- The LED lights *yellow* when the instrument is selected (and can be played from any connected external MIDI controller). You select an instrument by pressing its ***Instrument•Sequence Track*** button. Pressing the button a second time "de-selects" the instrument, changing the yellow LED back to red. If none of the LEDs are yellow, that means that no instruments are selected, and playing an external MIDI controller won't make any sound.
- The LED flashes *yellow* when the instrument is "Stacked" with one or more instruments. An instrument that is stacked will play simultaneously with any other instruments that are selected or stacked. You stack an instrument by pressing its button *twice in rapid succession* (or "double-click" on the ***Instrument•Sequence Track*** button, to borrow a term from those computers that use a "mouse"). In the illustration above, ***Instrument•Sequence Track 3*** is stacked with ***Instrument•Sequence Track 2***. You will hear both instruments when you play an external MIDI controller (wherever their keyboard ranges overlap, that is).

Note: Instruments can be played from an external MIDI controller without being selected, when Edit/System•MIDI, IN MODE= MULTI, or MONO B. See *Section 2 — System•MIDI* for details.

- Select a loaded instrument (one whose LED is red) by pressing its ***Instrument•Sequence Track*** button. The display now looks like this:



Once you have loaded one or more instruments, select the instrument you want to play by pressing the appropriate ***Instrument•Sequence Track*** button. The LOAD indicator stops flashing, and the ASR–10 shows you the name of the selected instrument, and its volume setting.

You can adjust the volume of the instrument with the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons. This lets you easily balance the levels of several instruments when splitting the keyboard or stacking sounds. If any other instruments are loaded, you can select them in the same way and adjust their volumes. Each time you select an instrument, its name and volume setting will appear in the display.

Hint: When TRANSMIT ON = INST CHAN, on the Edit/System•MIDI page, editing the VOLUME value will transmit MIDI Volume messages (Controller #7) on the selected ***Instrument•Sequence Track's*** MIDI OUT CHANNEL (on the Edit/Instrument page).

Hint: To adjust the volume of each loaded Instrument without selecting them, first select one instrument, and then use the ***Left/Right Arrow*** buttons to view the volumes of the other Instruments that are loaded.

- Try selecting different instruments. Try "double-clicking" to stack instruments. Click once on a selected or stacked instrument to de-select it.

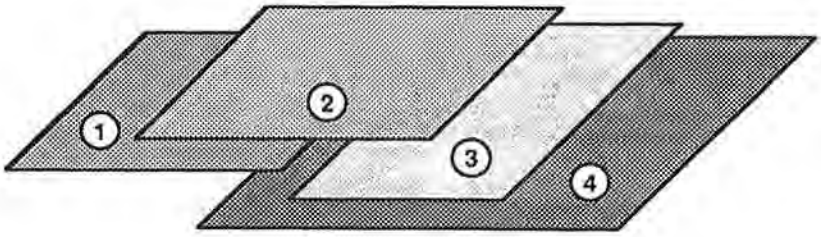
- Pressing the ***Load*** button again returns you to viewing disk files (LOAD flashing). You can also press ***Cancel•No*** to toggle back and forth between looking at disk files (LOAD flashing) and looking at instruments in the internal memory (LOAD indicator lit but not flashing).

Keyboard Range (How Instruments Share the Keyboard in Load Mode)

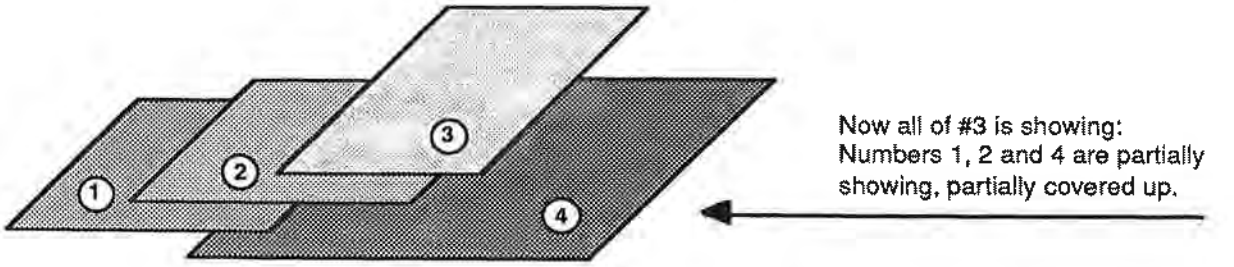
Whenever the ASR–10 is in Load Mode (which is the normal performance mode) you are not limited to one instrument on the keyboard at a time (or two, or three for that matter). Each instrument has a *Keyboard Range*, which is the section of the keyboard that it will occupy when selected.

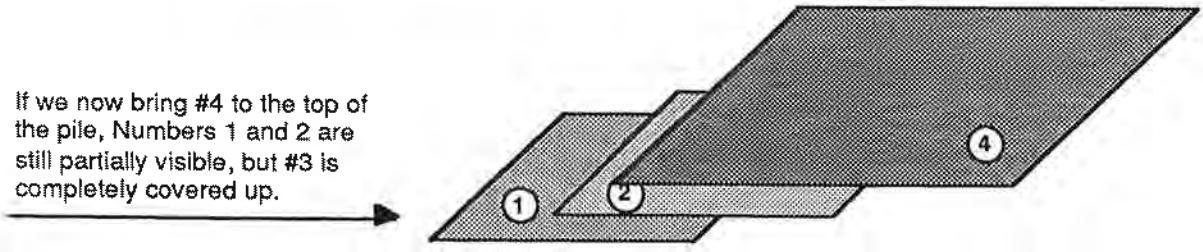
The range of an instrument can be as much as the full 127 keys of the MIDI Specification (for playing the ASR–10 from an external controller via MIDI) or as little as a single key. Up to eight instruments can inhabit the keyboard in a way that might be termed a "Pile o' Instruments."

Imagine four pieces of paper, each a different size, piled on top of each other:



In the picture above, sheet #2 is on top of the pile — all of it is showing. The other three sheets are partially covered up. Now suppose we could magically bring sheet #3 up to the top of the pile. The pile would look like this:





So it is with instruments loaded into the ASR–10 in Load mode. When you select an instrument, it is *brought to the top of the pile*. That is, you will hear the most recently selected instrument over its whole range when you play the keyboard. Any previously selected or stacked instruments will be "covered up" wherever their ranges overlap with that of the newly selected one.

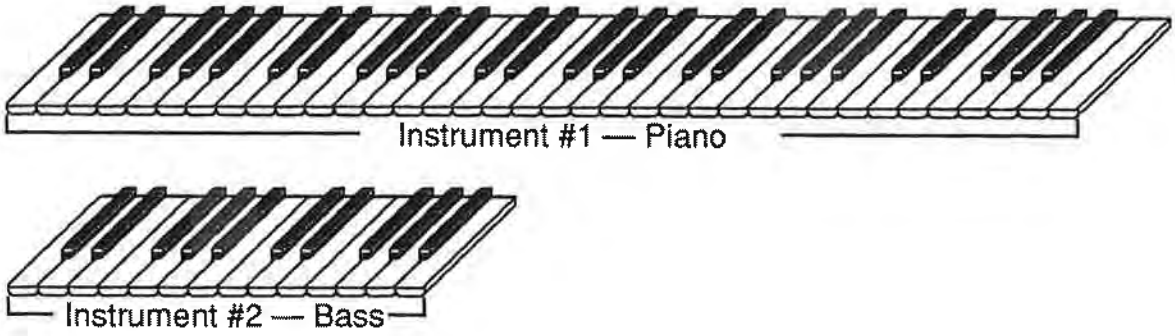
Wherever an instrument is not "covered up," it will continue to play on the keyboard. So you can see that "splitting" the keyboard is as easy as selecting two or more instruments with different (or overlapping) ranges.

- All instruments that are selected (LED lit yellow) are "on the pile," even if they are partially covered up by another instrument.

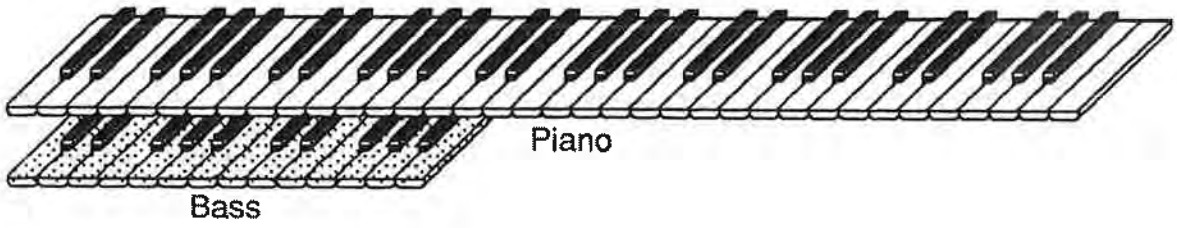
Now let's try the "sheets of paper" analogy in terms of instruments on the keyboard. Suppose you have two instruments loaded into the ASR–10:

- Instrument #1 is a piano sound, whose range is the entire keyboard;
- Instrument #2 is a bass sound, whose range is only the bottom two octaves.

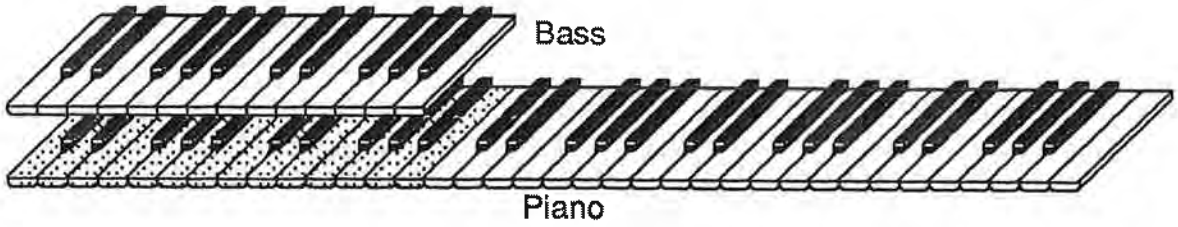
The ranges of the two instruments can be shown like this:



If you select the piano, it will play over the whole keyboard, covering up the bass entirely:



Now you select the bass. This brings it to the top of the pile, and it covers up the piano, but only in the region where their ranges overlap (the bottom two octaves):



If you play now, you will hear the bass in the bottom two octaves and the Piano over the rest of the keyboard. If you deselect the bass (by pressing its ***Instrument•Sequence Track*** button again) its LED goes from yellow back to red and it returns the whole keyboard range to the piano.

This same logic applies no matter how many instruments are loaded and selected. Whichever one you select comes to the "top of the pile."

Note: The "Pile" described here only exists in Load Mode. In Edit and Command modes, only *one* instrument can be selected at a time.

Stacking Instruments

With either the bass or the piano selected, double-click the other ***Instrument•Sequence Track*** button. Now in the lower two octaves you hear the bass *and* the piano.

Bass and Piano Stacked

Instruments that you "stack" by double-clicking their instrument buttons will play *simultaneously* with whatever other instruments are on the pile. Selecting another instrument (by pressing its ***Instrument•Sequence Track*** button and not by double-clicking), will bring that instrument to the top, covering up any stacked instruments as well as any selected ones. Up to all eight instruments can be stacked at once.

Additional Notes:

- Whenever a selected instrument is covered up completely by the selection of another, its LED changes from yellow back to red — it is automatically deselected.
- Up to all eight instruments can be selected and can play on some part of the keyboard at once, as long as none is covered up completely.
- You can reset any instruments key range. This is done on the Edit/Instrument page.

Loading Sequencer Data

There are two ways that ASR–10 Sequencer data can be stored on a disk:

- **SONG File.** A song file contains a song *and* all its related sequences. Loading a song file from disk will completely erase the current contents of the ASR–10 sequencer memory, replacing whatever is there with the song and sequences from the disk file.
- **SINGLE SEQUENCE File.** This type of file contains just one sequence. Loading a single sequence file will *not* erase the sequencer data already in memory — the new sequence simply becomes one more sequence in the internal memory which can be selected or used as a step in a song. There can be up to 80 sequences in memory at once.

Loading a Song or a Single Sequence from Disk

Both types of sequencer files are accessed from the Load/Seq•Song page.

- Press ***Load***, then press ***Seq•Song.***
  The LOAD indicator flashes, meaning that the ASR–10 is showing you disk files.
- Press the ***Up*** or ***Down Arrow*** button until the file you want to Load is showing on the display:

When a Song file is showing, the SONG indicator lights next to the SEQ indicator.

When a single Sequence file is showing, only the SEQ indicator lights.

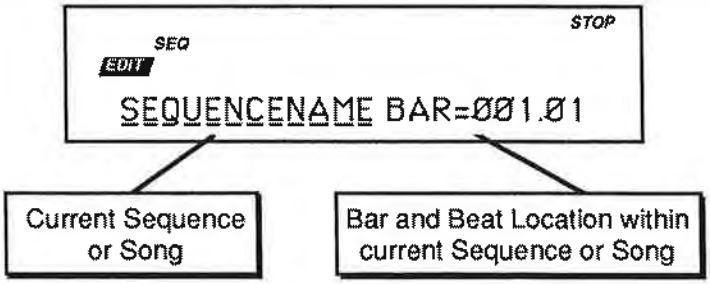
- Press ***Enter•Yes*** to load the sequencer file showing on the display. After you have loaded a song or sequence, it is automatically selected.

Important:

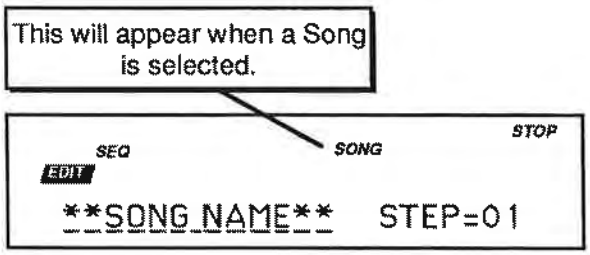
Remember that loading a song or sequence does not ensure that the right instruments are loaded into the proper ***Instrument•Sequence Track*** locations. Each track of each sequence will play whatever instrument is in that location at the time. The way to make sure everything is in the right place is to save the contents of memory (including the song) as a bank.

Selecting a Sequence/Song

Press the ***Edit*** button, then the ***Seq•Song*** button. The following screen will appear:



This screen is where you select a sequence or the song. With the selected sequence or song underlined (as shown above), use the ***Up/Down Arrow*** buttons or the ***Data Entry Slider*** to select a different sequence or song. You can always get back to this screen by double-clicking the ***Seq•Song*** button (when in Edit mode). The display will indicate whether a song is selected:



Playing a Sequence/Song

Try selecting a sequence, and pressing the ***Play*** button in the sequencer section, to the right of the display. The selected sequence will begin to play.

While one sequence is playing you can select another one. You will see the name of the new sequence in the display, but the original one will continue to play. When the first sequence is finished, the sequencer will switch to the new sequence, and it will play. In this fashion you can string sequences together in real time, as they play.

Press the ***Stop•Continue*** button or the Left Foot Switch to stop the current sequence.

Saving Sequencer Data

Saving a Single Sequence to Disk

Use the SAVE CURRENT SEQUENCE command to save a single sequence.

- On the Edit/Seq•Song page, select the sequence you want to save.
- Insert a formatted disk into the drive.
- Press ***Command***, then press ***Seq•Song.***
- Press the ***Left*** or ***Right Arrow*** button until the display reads SAVE CURRENT SEQUENCE.
- Press ***Enter•Yes.***
- Edit the sequence name (if needed):

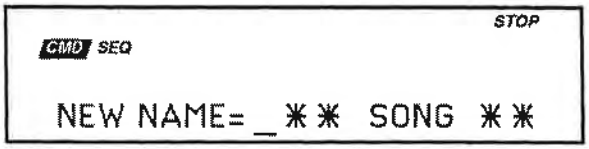
The display shows the current name of the sequence, with a cursor (underline) beneath the first character. If you want to give the sequence a new name, do so at this time. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the underlined character, then press the ***Left/Right Arrow*** buttons to move the underline. Repeat until the display shows the name you want (if you don't need to rename the sequence, just skip this step).

- Press ***Enter•Yes***. The display will read SAVING <FILE NAME> while the sequence is being saved.
- If there is already a sequence file with the same name on the disk, the display will ask DELETE OLD VERSION? Press ***Enter•Yes*** to save the sequence, replacing the one on the disk. Or press ***Cancel•No*** to abort the procedure.
- If there is not enough free space on the disk, the display will say NOT ENOUGH DISK SPACE. Save the sequence to another disk (or delete some files from the disk).
- If you try to SAVE CURRENT SEQUENCE while the song is selected, the ASR–10 will not execute the command, responding USE SAVE SONG + ALL. Again, make sure the sequence is selected before trying to save it.

Saving a Song (along with all Sequences) to Disk

Once you have created a song or made changes to an existing one, you can save the song to a formatted ASR–10 disk. In addition to saving the song itself, the SAVE SONG + ALL SEQS command saves all the individual sequences currently in memory (whether they are part of the song or not). To save a song:

- Insert a formatted disk into the drive.
- Press ***Command***, then press ***Seq•Song***.
- Press the ***Left*** or ***Right Arrow*** button until the display reads SAVE SONG + ALL SEQS.
- Press ***Enter•Yes***.
- Edit the song name (if needed):



The display shows the current name of the song, with a cursor (underline) beneath the first character. If you want to give the song a new name, do so at this time. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the underlined character, then press the ***Left/Right Arrow*** buttons to move the underline. Repeat until the display shows the name you want (if you don't need to rename the song, just skip this step).

- Press ***Enter•Yes***. The display will read SAVING <SONG NAME> while the song is being saved.
- If there is already a song file with the same name on the disk, the display will ask DELETE OLD VERSION? Press ***Enter•Yes*** to save the song, replacing the one on the disk. This is for updating songs to which you have made changes. Or press ***Cancel•No*** to abort the procedure.
- If there is not enough free space on the disk, the display will say NOT ENOUGH DISK SPACE. Save the song to another disk (or delete some files from the disk).
- Note that the song and all sequences in memory are saved as one file. You cannot extract a single sequence from a song file and load it separately. If you want access to a sequence individually, you must first load the song into the internal memory, and then save the individual sequence using the SAVE CURRENT SEQUENCE command.

Deleting a File from Disk

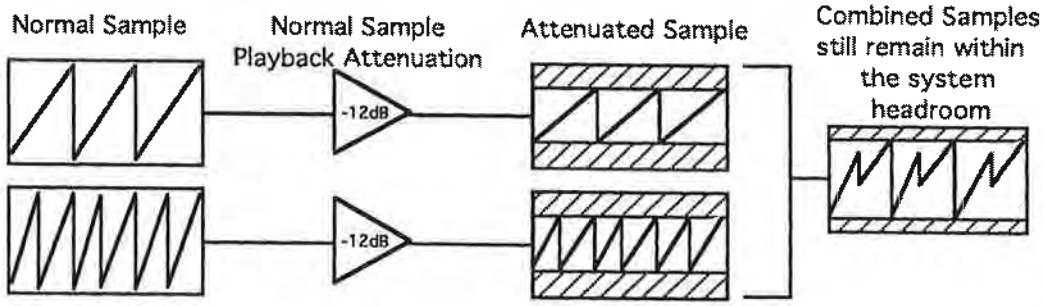
Deleting of disk files is done from Load mode. To delete (erase) a file from a disk:

- Press ***Load***, followed by the page button (***Instrument***, ***Seq•Song***, ***System•MIDI***, or ***Effects***) for the type of file you want to delete.
- Press the ***Up*** or ***Down Arrow*** button until the file you want to delete is showing on the display.
- While holding down the ***Load*** button, press ***Cancel•No***. The display will ask DELETE <FILE NAME>?
- Press ***Enter•Yes***. Press ***Cancel•No*** to abort this procedure.

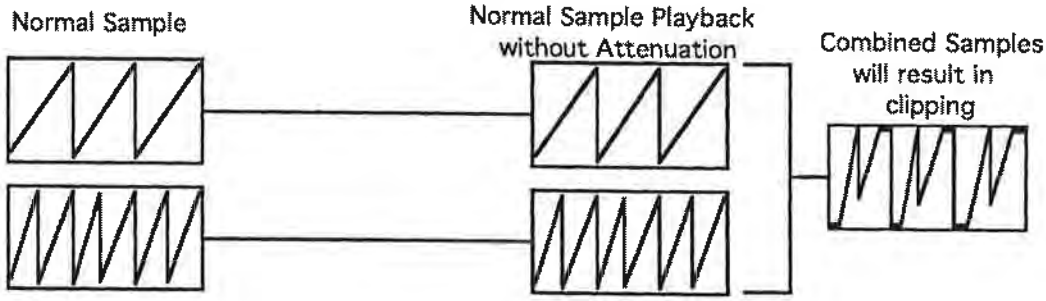
Managing Your System Headroom

The ASR–10's 16-bit stereo output converters provide 96dB of dynamic range. This figure of 96dB is referred to as the *system headroom.* Since the ASR–10 is capable of playing 31 notes of polyphony, this 96dB of system headroom must be divided amongst these 31 voices. If the output level of all sustaining voices exceeds 96dB, this will result in clipping. As a result, normal sample playback is attenuated by 12dB to allow multiple voices to sustain simultaneously without clipping, as shown below:

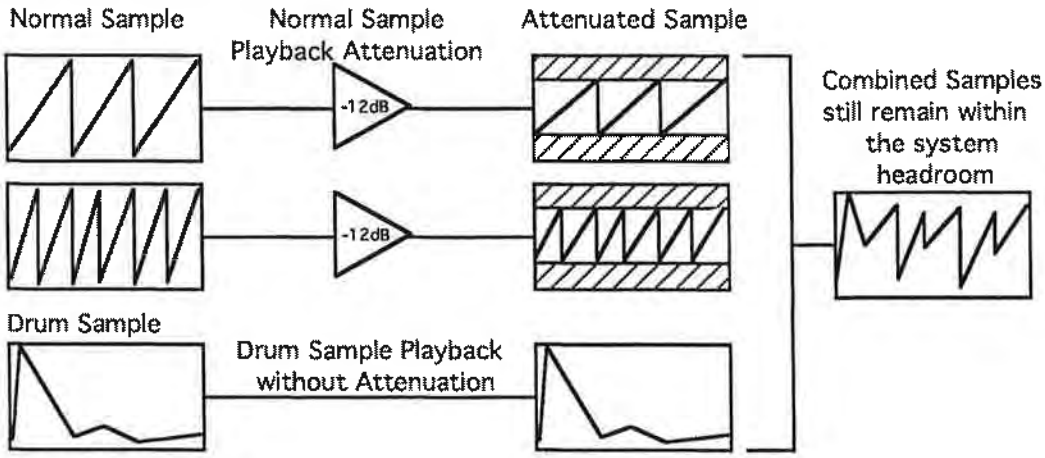
With Attenuation (BOOST=OFF)



Without Attenuation (BOOST=ON)



There may be times that this attenuated level does not offer enough perceived loudness for a certain type of sound. For example, percussion sounds only play at their peak volume for a brief instant before decaying quickly to a lower volume. As a result, it is often desirable to have that peak volume be even louder than the loudest attenuated level. This is the purpose for the Edit/Amp BOOST parameter. It provides a 12dB boost in volume to allow a WaveSample to use all 96dB of headroom. Since it is unlikely that two percussion sounds will both use the entire 96dB of headroom at the same exact instant, it is a relatively safe assumption that turning boost on for multiple percussion sounds will not result in clipping.



The BOOST parameter should not be turned on for all WaveSamples.

Sustaining sounds can remain at their peak volume for as long as a key is held down. As a result, if boost is turned on for a sustaining sound, it is *very* likely that playing multiple notes will result in clipping, even if the notes are not all started at the same time.

After sampling with REC SRC= MAIN-OUT, the Edit/Amp BOOST parameter is automatically set to ON for the new WaveSample. This is done because you have sampled a signal that was attenuated by 12dB and now the sample will be played back attenuated by another 12dB. Turning BOOST= ON, compensates for this effect. As a final step, after sampling the MAIN OUT, it is recommended that you optimize your use of headroom by setting the Edit/Amp BOOST= OFF, and performing the Command/Amp NORMALIZE GAIN command on the new WaveSample.

The key to managing your headroom is to determine which non-sustaining sounds will be the loudest, and set the BOOST parameter ON only for those sounds.

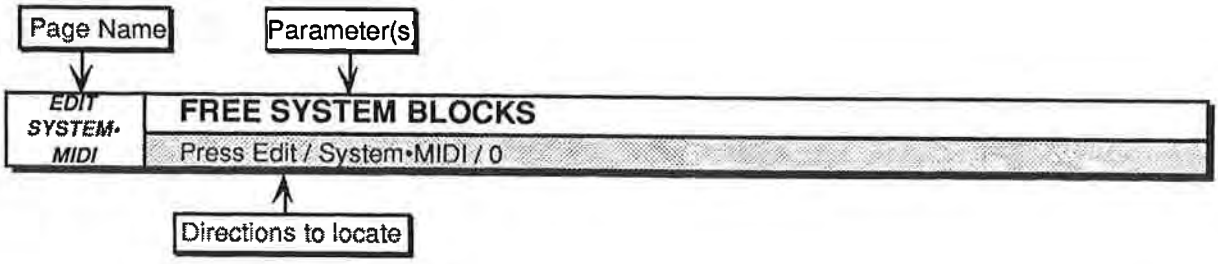
Audio Track Headroom

The Audio Tracks are permanently set to BOOST= ON. To prevent clipping, the Edit/Track MIX parameter for the Audio Tracks defaults to MIX= 60. When MIX= 99, the Audio Track will use the entire 96dB dynamic range. If you hear clipping when monitoring audio input on an Audio Track, adjust the Input Level trim control so that the red Input Level Peak LEDs only light occasionally (the red Peak LEDs light at 6dB below clipping), and then set the Edit/Track MIX parameter to a lower value.

Parameter Illustrations

The ASR–10 accesses parameters and commands through *Pages*. Each page is accessed by two button presses, the first being a Mode button and the second a Page button. Most parameters will also have a *Direct-Dial* number. This number can be pressed immediately after pressing the proper Mode and Page button to get to a specific parameter directly, bypassing scrolling with the ***Left/Right Arrow*** buttons.

When describing parameters or commands in this manual, parameter illustrations are used to communicate the location of the parameter and how to get there. For example:



These illustrations not only help to divide the manual into easy to locate sections for each screen, they also provide the following three important pieces of information about the parameter being discussed:

- Page Name — The page in which the parameter resides. In this case, it is the Edit/System•MIDI page.
- Parameter(s) — This tells you the name of the parameter or parameters as they're listed in that screen.
- Directions to Locate — This tells you how to get to the parameter screen. The directions will contain the Direct-Dial number for the parameter if one exists. If not, it will advise you to use the ***Left/Right Arrow*** buttons to scroll to the parameter.

System
MIDI

Section 2 — System•MIDI

These pages contain parameters and commands that affect the ASR–10 as a whole.

Edit/System•MIDI Page

This page contains global system and MIDI parameters. Due to the large number of parameters on this page, not all parameters have a direct dial number. Those that do will have that number listed in its parameter illustration. Parameters without direct dial numbers can be accessed with the ***Left/Right Arrow*** buttons.

Hint: By double-clicking the ***System•MIDI*** button in Edit mode, you can toggle between the first System Parameter and the first MIDI parameter.

Note: The settings for most of these parameters will be saved to the O.S. disk when you perform the SAVE GLOBAL PARAMETERS command (described later).

System Parameters

EDIT SYSTEM•MIDI	**FREE SYSTEM BLOCKS**
	Press Edit / System•MIDI / 0

This page shows the amount of available internal memory (in blocks).

EDIT SYSTEM•MIDI	**FREE DISK BLOCKS**
	Press Edit / System•MIDI / scroll using the arrow buttons

This page shows the amount of available memory (in blocks) on the disk currently in the drive (or on whatever SCSI Storage Device is selected as the storage device).

EDIT SYSTEM•MIDI	**MASTER TUNE**
	Press Edit / System•MIDI / 1

Adjusts the overall master tuning of the keyboard up or down as much as one semitone. A value of +0 will set the ASR–10 to concert A=440 tuning.

Range: -99 to +99 cents

EDIT SYSTEM•MIDI	GLOBAL BEND RANGE
	Press Edit / System•MIDI / scroll using the arrow buttons

Use this parameter to adjust the pitch bend range. The bend range is adjustable in one-semitone increments from 0 to 12. Each ASR–10 WaveSample can have its own bend range (which will override the global bend range) or can use this global bend range.

Range: 0 to 12 semitones

Note: When MASTER TUNE and GLOBAL BEND RANGE are edited, the ASR–10 will transmit the current value as a Registered Parameter Select message, on the MIDI BASE CHANNEL.

EDIT SYSTEM•MIDI	ALL NOTES OFF
	Press Edit / System•MIDI / scroll using the arrow buttons

This parameter determines whether the ASR–10 will respond to MIDI All Notes Off messages received via MIDI, on *any* MIDI channel. All Notes Off is a MIDI mode message that turns off all voices currently sounding at the time the message is received. It is not recognized when the ASR–10 is in OMNI mode.

OFF — The ASR–10 will ignore incoming MIDI All Notes Off messages.
ON — The ASR–10 will respond to incoming MIDI All Notes Off messages.

The ASR–10 does not *transmit* MIDI All Notes Off messages (though it will "pass on" any such messages via its MIDI Thru jack).

EDIT SYSTEM•MIDI	PEDAL (VOLUME / MOD)
	Press Edit / System•MIDI / 2

The optional CV Pedal (model CVP-1) can be used as a volume pedal or as a modulator. When PEDAL= VOLUME, the pedal will only affect the volume of the selected ***Instrument•Sequence Track(s)***, and it will be transmitted via MIDI as MIDI controller #7. When PEDAL=MOD, it will affect any modulation destination that has PEDAL selected as a modulator, and it will be transmitted via MIDI as MIDI controller #4. When PEDAL= VOLUME the ASR–10 will still receive PEDAL (MIDI controller #4) messages via MIDI.

Note: Some Yamaha™ TX modules (depending on the software version) receive MIDI controller #4 as volume. If PEDAL=MOD MIDI=4, and the pedal is not plugged in, the MIDI controller #4 value transmitted will default to 0, which could shut down the volume of the TX.

- When set to PEDAL=VOLUME MIDI=7 the optional CV Pedal acts as a volume pedal for the selected ***Instrument•Sequence Track(s)***.
- When set to PEDAL=MOD MIDI=4 the optional CV Pedal acts as a modulation pedal, and will cease to act as a volume pedal.

EDIT SYSTEM•MIDI	LEFT FOOT SW
	Press Edit / System•MIDI / scroll using the arrow buttons

When the optional SW-10 Dual Foot Switch is plugged into the rear panel Foot Switch jack, this page allows the Left Foot Switch to perform a variety of assignable functions. The Right Foot Switch is permanently dedicated to function as the Sustain pedal.

- OFF — This setting makes the ASR–10 ignore the Left Foot Switch. If you are using the single foot switch (SW-2) which came with the ASR–10, then you should keep this parameter set to OFF (the SW-2 will only function as a sustain pedal).
- FX MODSRC — This setting allows the left foot switch to function as an effects modulation source, assignable to any modulation destination in any effect.
- SAMPL YES — This setting allows you to initiate sampling with the Left Foot Switch. After pressing the ***Sample•Source Select*** button, successive presses of the Left Foot Switch will behave as though the ***Enter•Yes*** button has been pressed.
- STOP/CONT — When set to STOP/CONT, the left pedal of the SW-10 duplicates the action of the ***Stop•Continue*** button. When in standby record, pressing the Left Foot Switch will initiate recording.

Note: If you want the SW-10 to duplicate the behavior of the Patch Select buttons, plug it into the Patch Select jack located on the rear panel.

EDIT SYSTEM•MIDI	AUTO-LOOP FINDING (ON/OFF)
	Press Edit / System•MIDI / 3

This parameter enables or disables the "Expert-System" auto-loop function. If the auto-loop function is enabled, the ASR–10 will automatically go to zero crossings (with matching wave direction) whenever you move the loop start or end pointer. This is a very useful feature because finding zero crossings is essential to creating glitch-free loops. If the auto-loop function is disabled, the loop start and end parameters will move in one-sample increments.

Note: AUTO-LOOP FINDING will not function on a WaveSample that exists in a stereo layer (i.e. a layer in which STEREO LAYER LINK= ON).

MIDI Parameters

Few developments in recent years have had as great an impact on the way we make music as has the emergence of MIDI. Whether you are simply linking two keyboards together, playing a synth from a guitar controller, or driving a rack of samplers from a drum pad controller, MIDI makes it all possible. The evolution of MIDI has facilitated the merging of existing technologies and has inspired the creation of new technologies. ENSONIQ has always been an industry leader in MIDI development, and the ASR–10 embodies the latest advances in a state-of-the-art sequencer, controller keyboard, and multi-timbral sound generator.

EDIT SYSTEM•MIDI	**MIDI BASE CHANNEL**
	Press Edit / System•MIDI / 5

Selects the Base Channel on which the ASR–10 transmits and receives MIDI messages. The base channel is used to receive MIDI data while the ASR–10 is in POLY and MONO A modes. This is the MIDI channel that the ASR–10 will transmit on when Edit/System•MIDI, TRANSMIT ON=BASE CHANNEL. System Exclusive messages are always sent and received on the base channel.

Any of 16 MIDI channels may be selected as the base MIDI channel of the ASR–10. The effect of setting the base channel varies depending on the setting of the TRANSMIT ON= parameter, the MIDI Mode and whether data is being transmitted or received.

Receive — In POLY mode, keys, controllers, and program changes are only recognized if received on the base channel. In MONO A mode, program changes are received only on the base channel. The base channel is also used in MONO A mode as the first channel of the 8 channel range.

Transmit — The ASR–10 has two different MIDI transmitting schemes, depending on the setting of the TRANSMIT ON= parameter. When TRANSMIT ON=BASE CHAN, the ASR–10 always transmits only on the base channel. When TRANSMIT ON=INST CHAN, the ASR–10 always transmits on the Edit/Instrument, MIDI OUT CHANNEL for each Instrument.

Range: MIDI channels 1 through 16.

EDIT SYSTEM•MIDI	**TRANSMIT ON (INST CHAN/BASE CHAN)**
	Press Edit / System•MIDI / 6

- When set to INST CHAN, each of the eight ASR–10 ***Instrument•Sequence Tracks*** will transmit on its own MIDI channel (or not at all, if Edit/Instrument, MIDI STATUS=LOCAL). Each ***Instrument•Sequence Tracks***will have its own MIDI program number and its own PRESSURE setting.
- When set to BASE CHAN, the ASR–10 will always transmit notes, controllers and program changes on the Base MIDI channel and *only* on that channel, no matter what Instrument•Sequence Tracks are selected or stacked.

EDIT SYSTEM•MIDI	**BASECHAN PRESSURE**
	Press Edit / System•MIDI / scroll using the arrow buttons

When TRANSMIT ON=BASE CHAN, the ASR–10 will transmit either Key or Channel pressure or no pressure at all, depending on the setting of this parameter. When TRANSMIT ON=INST CHAN, this setting is ignored and each instrument transmits pressure according to its own Edit/Instrument PRESSURE MODE setting.

- When BASECHAN PRESSURE=OFF, no pressure will be transmitted via MIDI.
- When BASECHAN PRESSURE=KEY, the ASR–10 will transmit the most expressive type of pressure — key pressure via MIDI. The ASR–10's Poly-Key™ Pressure lets you modulate each note independently. If you press down on any given key within a chord, only that note will be affected by pressure—all other notes remain unmodulated.
- When BASECHAN PRESSURE=CHAN, the ASR–10 will transmit the most common type of pressure—channel pressure, via MIDI. With channel pressure, after a note is played, pressing down harder on the key will modulate *every* note currently playing. Like a mod wheel, channel pressure is "global," it affects the entire keyboard when activated.

You can consult the MIDI implementation chart of the MIDI device you will be using with the ASR–10, to see which, if any, type of pressure it responds to. Set this parameter to the appropriate value when playing or sequencing a particular device from the ASR–10.

EDIT SYSTEM•MIDI	**MIDI IN MODE**
	Press Edit / System•MIDI / 7

This parameter determines how MIDI information will be *received* by the ASR–10 and its various Instruments. MIDI In Mode has no effect on what MIDI information is transmitted.

There are five MIDI modes implemented in the ASR–10:

- OMNI — In this mode the ASR–10 will receive on all 16 MIDI channels. This mode is useful when you are only using a few instruments, and you are not concerned with setting up different channels for each device.
- POLY — In this mode the ASR–10 will receive only on the Base MIDI channel. MIDI information on all other channels will be ignored.
- MULTI — An ENSONIQ innovation, MULTI mode is the key to unlocking the potential of the ASR–10 as a multi-timbral receiver from an external MIDI sequencer. In MULTI mode, the 8 tracks of the current song or sequence, and the 8 individual instruments, can receive MIDI information independently and polyphonically on up to 8 different MIDI channels.
- MONO A — This mode is optimized for use with a MIDI guitar controller.
- MONO B — This mode is optimized for use with a MIDI guitar controller, with the added benefit of assigning a different instrument to each string.

Note: In OMNI, POLY, and MONO A modes, incoming MIDI will only play thosed ***Instrument•Sequence Tracks*** that are "Selected" or "Stacked."

Note: In MULTI and MONO B modes, independent of what ***Instrument•Sequence Tracks*** are "Selected" or "Stacked" on the front panel, the sounds you will hear when notes are received via MIDI will depend entirely on what MIDI channel(s) the MIDI data is received on.

Note: In MULTI and MONO B modes, different MIDI receive channels must be selected for each ***Instrument•Sequence Track*** that you want to receive via MIDI. The MIDI receive channel for each ***Instrument•Sequence Track*** is set with the Edit/Track, MULTI-IN MIDI CHAN parameter. If more than one sequence track is set to the same MULTI IN MIDI CHAN, only the lowest numbered ***Instrument•Sequence Track*** will receive via MIDI.

Note: In MULTI and MONO B modes, notes played on the ASR–10's own keyboard are not affected by incoming MIDI Controller messages. Only the ASR–10's own controllers will affect notes played on the ASR–10's keyboard.

MONO Modes

MONO modes are particularly useful for driving the ASR–10 from a guitar controller, or any other application where having up to eight independent, monophonic channels is desirable. The ASR–10 offers two types of MONO mode operation:

- MONO A is the mode you will probably use most often. This is the mode to use when you want to play the same sound on all the strings of your guitar controller. In MONO A, the ASR–10 receives on eight consecutive MIDI channels (the MIDI BASE CHANNEL through MIDI BASE CHANNEL +7) and will play whatever ***Instrument•Sequence Track***(s) are selected or stacked on the front panel. In other words, the ASR–10 behaves as it does in POLY or OMNI modes, except that it receives monophonically on multiple MIDI channels. This provides the advantage of multiple tracks that will respond independently to controllers received on multiple channels, but you do not have to set up the instruments for each track separately.
- MONO B is the mode to use if you want to be able to play a different sound on each string of your guitar controller. In MONO B, each ***Instrument•Sequence Track*** receives on its own Edit/Track MULTI-IN MIDI channel. In other words, the ASR–10 behaves as it does in MULTI mode, except that each ***Instrument•Sequence Track*** receives monophonically on multiple MIDI channels. This is the only way to get a different sound on each string when using a MIDI guitar controller.

Global Controllers in MONO A and B Modes

Global controllers are controllers sent on one channel that affect all other channels simultaneously. They can be useful in reducing the number of MIDI events required to achieve particular effects, and can thereby reduce the delays sometimes associated with overloading MIDI. Some guitar controllers can transmit global controllers, and the ASR–10 can respond to them.

In MONO mode (A or B) the base channel minus one becomes the MIDI channel for global controllers (pitch bend, pressure, etc.). For example, if the base channel is channel 3, any controllers received on channel 2 will be interpreted as global controllers and will affect *all* voices being played. If the base channel is channel 1, channel 16 becomes the channel for global controllers. Each track will also respond independently to controllers sent on its own channel. For example, each guitar string on a MIDI guitar can send independent pitch bend, while the "whammy bar" controller could be sent on the global channel to affect all voices.

Note: The ASR–10 will not receive note data via MIDI on the base channel minus one in MONO A and B modes. Therefore, we recommend that the MIDI BASE CHANNEL parameter on the Edit/System•MIDI page be set to the same channel as the ***Instrument•Sequence Track*** with the lowest numbered Edit/Track MULTI-IN MIDI CHAN number.

For more information on using MULTI and MONO modes, see *Section 17 — Sequencing/MIDI Applications*.

EDIT SYSTEM•MIDI	**MIDI CONTROLLERS (ON/OFF)**
	Press Edit / System•MIDI / scroll using the arrow buttons

This determines whether the ASR–10 will send *and* receive MIDI controllers — pitch bend, mod wheel, pressure, volume, sustain, etc. When OFF, the ASR–10 will not transmit or receive MIDI controllers.

EDIT SYSTEM•MIDI	**MIDI SYS-EX (ON/OFF)**
	Press Edit / System•MIDI / scroll using the arrow buttons

This parameter determines whether the ASR–10 will send *and* receive MIDI System Exclusive messages, such as remote programming instructions or WaveSample dumps. When OFF, the ASR–10 will neither transmit nor receive System Exclusive messages.

Note: This parameter does not have to be ON for the ASR–10 to record and store Sys-Ex messages from external devices using the MIDI SYS-EX RECORDER function on the Command/System•MIDI page.

EDIT SYSTEM•MIDI	**MIDI PROG CHANGE (ON/OFF)**
	Press Edit / System•MIDI / 8

This parameter enables (ON) or disables (OFF) ASR–10 reception of program changes via MIDI (the default is ON).

To Send a MIDI Program Change from the ASR–10:

- Hold down an ***Instrument•Sequence Track*** button.
- *While holding down the* ***Instrument•Sequence Track*** *button*, "type in" the number (from 1-128) of the desired program change on the numeric keypad buttons of the ASR–10; then
- Release the ***Instrument•Sequence Track*** button. When the button is released, the program change is sent.

If the ASR–10 has been set to TRANSMIT ON= INST CHAN, the program change will be sent on the MIDI channel of the instrument whose button you held down while typing in the number. If it is set to TRANSMIT ON= BASE CHAN, the program change will always be sent on the base channel.

MIDI program changes can also be sent by setting TRANSMIT ON=INST CHAN and then editing the Edit/Instrument, MIDI OUT PROGRAM value.

When ***Instrument•Sequence Tracks*** are selected in solid LOAD mode, the ASR–10 will transmit the MIDI OUT PROGRAM number for the selected ***Instrument•Sequence Track(s).***

Receiving Program Changes

What the ASR–10 does when it receives a program change depends on the current MIDI IN MODE.

If the MIDI IN MODE = OMNI, POLY, or MONO A, incoming program changes 1-24 will *Select*, *Deselect,* or *Stack* the eight ***Instrument•Sequence Tracks*** of the ASR–10 in the following fashion:

- Program changes 1-8 will *Select* ASR–10 Instruments 1-8 (LED will light yellow), if received by an instrument which is not selected.
- Program changes 1-8 will *Deselect* Instruments 1-8 (LED will change to red), if received by an instrument which is already selected.
- Program changes 9-16 will *Stack* Instruments 1-8 (LED will flash yellow).
- Also, program changes 17-24 will select Performance Presets 1-8.

If the MIDI IN MODE = MULTI or MONO B, incoming program changes are used to load Instruments as well as Banks into ***Instrument•Sequence Track*** locations and to enter and exit directories. To load a given file into an ***Instrument•Sequence Track*** location, press the ***Load*** button, and send the ASR–10 a program change that is the file number +1 on the ***Instrument•Sequence Track's*** Edit/Track MULTI-IN MIDI CHANNEL. To load a Bank file, send the ASR–10 a program change that is the Bank file number +1. When a program change is received, the ASR–10 will load the appropriate file from the currently selected storage device (FLOPPY, SCSI 0-7). Incoming MIDI program changes 101-128 will invoke Macros 0-27.

Note: In MULTI and MONO B modes, program change 1 has a special function, which is to EXIT TO the previous directory. This is particularly useful for moving around a SCSI Storage Device from a remote MIDI controller or computer.

EDIT SYSTEM•MIDI	**MIDI SONG SELECT (ON/OFF)**
	Press Edit / System•MIDI / scroll using the arrow buttons

When this parameter is ON, a remote sequencer or drum machine can be instructed to select a new song when you load a new song into the ASR–10, or when the currently loaded song is selected (the default is ON). The song can be assigned a MIDI Song Select number from 0 through 127. MIDI Song Select numbers 0 through 127 are transmitted via MIDI when you load a new song into the ASR–10, or when the currently loaded song is selected.

When the song is selected, MIDI Song Select numbers can be assigned as follows:
- Press ***Command***, then ***Seq•Song***. Scroll to SEQUENCER INFORMATION.
- Press ***Enter•Yes***. Scroll left until the display reads MIDI SONG SELECT=##.
- Select a MIDI Song Select number 0 through 127 using the Data Entry Controls.

MIDI Song Select numbers are stored with the song when you save the song to disk.

EDIT SYSTEM•MIDI	MIDI XCTRL NUMBER
	Press Edit / System•MIDI / 9

Range: 0 to 127

Most controllers on a synthesizer — mod wheel or foot pedal, for example — have a MIDI controller number that can be assigned to this parameter. Doing so will make a particular external controller available as a modulator to any of your programs.

One modulation source that can be selected is XCTRL (external controller). When the ASR–10 receives MIDI Controller messages corresponding to this controller number, they will be routed to all parameters that have been programmed with XCTRL as a modulation source. Suppose, for example, you are playing the ASR–10 from a keyboard with a breath controller (or want to use a breath controller as a modulator when playing the ASR–10 keyboard). You can set up a WaveSample on the ASR–10 in which the filter cutoff frequency is modulated by XCTRL. If you then set "XCTRL=02," the breath controller can modulate the filter, or any other parameter that has its modulation source set to XCTRL.

The following controller numbers are commonly supported:

Number	Controller	Number	Controller	Number	Controller
1	Modulation Wheel	10	Pan	71	Harmonic Content (Timbre)
2	Breath Controller	64	Sustain	72	Release Time
4	Foot (Pedal) Controller	66	Sostenuto	73	Attack Time
6	Data Entry	70	Sound Variation (Patch Selects)	74	Brightness
7	Volume				

Although the range of this control is from 0 to 127, many of the values other than those listed above have no "approved" function, yet. They exist to provide flexibility and to accommodate future MIDI standards.

EDIT SYSTEM•MIDI	MULTI CONTROLLERS
	Press Edit / System•MIDI / scroll using the arrow buttons

This parameter determines whether local controllers (such as the patch select controllers, volume pedal, etc.) will affect all tracks or only the selected track(s). Setting this to ON is useful if you are playing the ASR–10 from an external drum pad or guitar controller in MULTI or MONO B modes and want local controllers to affect all the tracks.

- ON — Local controllers will affect all ***Instrument•Sequence Tracks*** when the ASR–10 is in MULTI or MONO B modes.
- OFF — Local controllers will affect only the ***Instrument•Sequence Tracks*** selected or layered from the front panel, no matter what is being received via MIDI.

EDIT SYSTEM•MIDI	ENTER PLAYS KEY
	Press Edit / System•MIDI / scroll using the arrow buttons

This parameter determines the note value (A0 to C8) that pressing ***Enter•Yes*** on this screen (or on the Edit Context page) will play. In other words, pressing ***Enter•Yes*** on either the Edit Context page, or on this screen will be the equivalent of playing a note on the controller keyboard, at a velocity of 127. This provides a way to manually trigger notes from the front panel of the ASR–10 Module without a MIDI Controller. See *Section 16 — Instrument Programming Applications* for more information.

Command/System•MIDI Page

Due to the large number of commands on this page, not all commands have a direct dial number. Those that do will have that number listed in its command illustration. Commands without direct dial numbers can be accessed with the ***Left/Right Arrow*** buttons.

CMD SYSTEM•MIDI	**FORMAT FLOPPY DISK**
	Press Command / System•MIDI / 0

Before you can save ASR–10 instruments, banks, or sequences to a disk it must be formatted. You can use a blank disk or one that was previously formatted by a different device, such as a computer. Note that formatting will completely erase anything on the disk.

- Insert the disk to be formatted into the drive.
- Press ***Command***, then ***System*** and scroll to FORMAT FLOPPY DISK.
- Press ***Enter•Yes***. The display will read DISK LABEL=DISK000
  Use the ***Data Entry Slider*** and the ***Arrow*** buttons to give the disk a unique name by which you can identify that disk. This is very *important*, as the ASR–10 will ask you for this disk by name if an instrument or song on the disk is used in a bank. Also, make sure you write the Disk Label name on the outside of each disk after you are done formatting it. After assigning a Disk Label, you can change the default prompt from DISK000 to the last label you assigned by using the "SAVE GLOBAL PARAMETERS" command (discussed later in this section). In doing this, you can ensure that all of your personal disks will have the same first four characters.
- Press ***Enter•Yes***.
  The display will read FORMAT TYPE. Here you select which type of formatting you would like to use to format your floppy disk:

ENSONIQ — this standard format allows the ASR–10 to read the disk, and offers the largest amount of free blocks (DSDD disks format to 800k and DSHD disks format to 1600k).

COMPUTER — this unique format provides a lower disk capacity, but by having the first sector labelled sector 1, allows MAC and IBM computers that are running the appropriate translation software to read the disks.

- After choosing a format type, press ***Enter•Yes***.
  The display will read ERASE AND FORMAT DISK?
- Press ***Enter•Yes***.

While the disk is being formatted, the display reads FORMATTING... When it's done the display says FORMAT COMPLETE. If the format fails, the display will read FORMAT FAILED. Try again with a different disk.

Note: After formatting a disk you can, if you want, copy the ASR–10 Operating System (O.S.) onto that disk (see below). Having the O.S. on your sound disks can be convenient. However, putting the O.S. on a disk leaves a little less space for saving instruments and sequences.

CMD SYSTEM•MIDI	COPY O.S. TO DISK
	Press Command / System•MIDI / 1

Use this command to put the ASR–10 Operating System (O.S.) on a new disk that you have formatted, or to update the Operating System on an existing disk with a newer version.

Note: The Operating System can only be copied to a blank disk, or a disk (with or without sounds and sequences) that has an existing operating system on it. You cannot copy the Operating System to a disk that only has instruments or sequences. Attempts to do so will result in an error message.

This procedure requires erasing whatever is in the ASR–10 Internal Memory, so make sure that any important data has been saved to disk. Also, you'll need a disk containing the Operating System that you want to copy.

- Insert the disk containing the O.S. you want to copy (the source disk) into the drive.
- Select COPY O.S. TO DISK.
- Press ***Enter•Yes***.
  The display says MUST ERASE MEMORY, OK? If you need to save any sounds or sequences, press ***Cancel•No*** and save the data before proceeding.
- Press ***Enter•Yes***.
  The display says INSERT MASTER O.S. DISK. Make sure the source O.S. disk is in the drive.
- Press ***Enter•Yes***.
  The display says READING O.S. INTO MEMORY, and then INSERT FORMATTED DISK.
- Insert the disk onto which you want to copy the O.S. (the destination disk) into the drive.
- Press ***Enter•Yes***.
  The display reads WRITING O.S. TO DISK while the O.S. is being copied to the disk.

When it's done, the display reads COPY O.S. DONE. ANOTHER? If you want to copy the same O.S. to another disk, insert another formatted disk and press ***Enter•Yes***. You can repeat this procedure as many times as you like. This makes it easy to update all your O.S. disks when a new ASR–10 Operating System is released.

- When you are done, press ***Cancel•No***.

CMD SYSTEM•MIDI	SAVE GLOBAL PARAMETERS
	Press Command / System•MIDI / 2

The SAVE GLOBAL PARAMETERS Command saves the settings for the following parameters:

- All Edit/System•MIDI parameters
- All Sample•Source Select parameters
- The current FX Select•FX Bypass page setting
- All Edit (audio) Track parameter settings
- The MULTI-IN MIDI CHAN settings on the Edit/Track page
- The Edit/Seq•Song CLOCK SOURCE, CLICK settings, and SEQ COUNTOFF mode
- The Source Monitor status for the audio tracks

These parameters are global to the ASR–10 and are not saved with the instruments, sequences, or banks. For example, if you have a firm playing style, it would be convenient to set the ASR–10 always to respond to your particular playing style. You would set TOUCH=FIRM 4 on the Edit/System•MIDI page, then perform the SAVE GLOBAL PARAMETERS command to a disk containing the current Operating System.

- Select SAVE GLOBAL PARAMETERS. Make sure you have a formatted, non-write-protected disk containing the most recent Operating System in the disk drive.
- Press ***Enter•Yes***. The display reads DISK COMMAND COMPLETED.

The next time you "boot" the ASR–10 with that disk, the new settings will be loaded in as the defaults.

Hint: If you use several different MIDI reception configurations in your studio, make several copies of the O.S. disk, and save different GLOBAL PARAMETERS on each one. You can then use the LOAD GLOBAL PARAMETERS command (Command/System•MIDI page) to load the different configurations from each O.S. disk.

CMD SYSTEM•MIDI	LOAD GLOBAL PARAMETERS
	Press Command / System•MIDI / 3

This command allows you to load global parameters from disk, without having to reboot the ASR–10.

- Select LOAD GLOBAL PARAMETERS. Make sure you have the disk containing the global parameters and operating system loaded into the disk drive.
- Press ***Enter•Yes***. The display reads DISK COMMAND COMPLETED.

CMD SYSTEM•MIDI	CREATE DIRECTORY
	Press Command / System•MIDI / 4

Use this command to create a new directory on the current disk. Directories are not generally necessary when working with floppy disks, as it is unlikely that you will fit more than 38 files on a floppy. However, a hard disk connected to the SCSI Interface can hold hundreds of files, and directories are a necessary part of managing and organizing those files.

CMD SYSTEM•MIDI	CHANGE STORAGE DEVICE
	Press Command / System•MIDI / 5

Use this command to determine whether the ASR–10 will use floppy disk or an external SCSI Storage Device.

- Select CHANGE STORAGE DEVICE. Press ***Enter•Yes***.
- Select LOAD DEVICE=FLOPPY/SCSI 0 through 7.
- You **must** press ***Enter•Yes*** to change the storage device. Otherwise, you will get the COMMAND ABORTED message.
- Press ***Enter•Yes***. The display reads COMMAND COMPLETED.

CMD SYSTEM•MIDI	SAVE MACRO FILE
	Press Command / System•MIDI / 6

Macros provide a way to move quickly from one place in the directory structure to another. They are primarily for use with a SCSI Storage Device.

- Press ***Enter•Yes***. The display will read FILENAME= MACROFILE 1.
  Use the ***Data Entry Slider*** and the ***Arrow*** buttons to name the macro file. This will help you to identify your files quickly. After you've named your macro file,
- Press ***Enter•Yes***.
  The display will read SAVING (macro file name), then it will show COMMAND COMPLETED.

See **Macros** later in this section for more information.

CMD SYSTEM•MIDI	COPY FLOPPY DISK
	Press Command / System•MIDI / 7

The COPY FLOPPY DISK command lets you duplicate the contents of one entire disk (the *source disk*) onto another disk (the *destination disk*). This function will only work if both disks are the same size (**both** High Density disks or **both** Double Density disks). It is a good practice to regularly back up your valuable data — including the O.S. disk that came with the ASR–10 — in this way. You can use either blank or formatted disks. During the COPY FLOPPY DISK command, the ASR–10 will ask you if you want to format the blank disk. Make sure the formatted destination disks do not contain any files that you want to keep because they will be erased. It is also a good idea to *write-protect* the source disk, so there's no chance of losing data.

Note: This procedure uses the internal memory to temporarily hold the information while copying it between disks. The fewer files you have loaded in the internal memory before you start, the fewer times you will have to swap disks during the copy procedure. You will only have to swap disks if the size of the source disk you are copying is greater than the amount of free internal memory.

To Make a Backup Copy of a Disk:

- Slide open the plastic write-protect tab on the source disk (the original) so that the disk is write protected (you can see through the hole). This is an extra precaution to safeguard the data.
- From the COPY FLOPPY DISK command, press ***Enter•Yes***. The display shows INSERT SOURCE DISK (the one you want to copy).
- Insert the source disk, then press ***Enter•Yes***.
- The drive will engage and the display will flash READING SOURCE DISK. Once the drive stops, the display will change to show INSERT DEST DISK.
- Insert the destination disk (the one you want to copy to), and then press ***Enter•Yes***. If the destination disk is unformatted, the display will ask ERASE AND FORMAT DISK? Press ***Enter•Yes*** to format the disk. When formatting is complete, the drive will engage and the display will flash WRITING DEST DISK.
- After writing to the destination disk, the display will read VERIFYING DEST DISK. If the copy is complete, then the display will read DISK COMMAND COMPLETED.
- If the size of the source disk you are copying is greater than the amount of free internal memory, the display will change to show INSERT SOURCE DISK again. Repeat the preceding four steps as prompted by the ASR–10 display until the display

Note: When a disk is copied, the Disk Label Number is copied from the source disk to the destination disk.

COPY FLOPPY messages:

- DISK WRITE-PROTECTED — When asked for the destination disk you inserted a write-protected disk. The destination disk must have the write-protect tab closed (so you can't see through the hole).
- DISK IS NOT SOURCE DISK or DISK IS NOT DESTINATION DISK — You put the wrong disk in the drive when prompted for a certain disk. This is not fatal; it doesn't abort the copy procedure. Just insert the requested disk and proceed.

CMD SYSTEM•MIDI	MIDI SYS-EX RECORDER
	Press Command / System•MIDI / 9

The ASR–10 can be used to receive, store, and send MIDI System Exclusive (Sys-Ex) messages. These Sys-Ex messages can be saved to and loaded from disk. *Since the ASR–10 erases the Internal Memory to "buffer" incoming Sys-Ex signals, you should make sure that any important data has been saved to disk before performing this command.*

The MIDI Sys-Ex recorder in the ASR–10 lets you store to disk things like:
- The program (patch) memory of any MIDI synth,
- Patterns from a drum machine,
- The sequence memory of a MIDI sequencer,
- Sample data dumps from samplers and sampling drum machines, and
- The preset memory of any MIDI effects device.

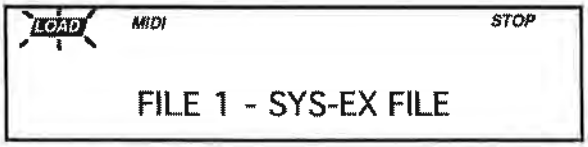
SAVING System Exclusive Data from an External Device
- Connect the MIDI Out of the sending device to the MIDI In of the ASR–10.
- Make sure that the ASR–10 O.S. disk is in the disk drive.
- Select MIDI SYS-EX RECORDER.
- Press ***Enter•Yes***. The display reads MUST ERASE MEMORY OK?
- Press ***Enter•Yes***. The display reads WAITING... The ASR–10 is ready to record any System Exclusive message that it receives.
- Send the Sys-Ex data from the external device to the ASR–10. The display will flash RECEIVING while the Sys-Ex data is being received by the ASR–10. If the Sys-Ex data have been received successfully, the ASR–10 will display CANCEL=QUIT ENTER=SAVE.

At this point, you can send the ASR–10 another Sys-Ex message (from a different device, for example) which will be stored right after the first one. You can save as many different messages as memory permits in a single Sys-Ex block. Each time a message is received, the display will flash RECEIVING and then return to CANCEL=QUIT ENTER=SAVE. When the data is later retransmitted, all the messages will be sent out in the order they were received. In this way you could load new data into all your devices with a single Sys-Ex Load command from the ASR–10.

To save the Sys-Ex file to disk:
- Insert a formatted disk into the ASR–10.
- Pressing ***Enter•Yes*** displays FILENAME=SYS-EX FILE. You can name your Sys-Ex file by using the data entry controls, if you wish.
- Press ***Enter•Yes***. The display reads SAVING <filename>, then DISK COMMAND COMPLETED.
- Pressing ***Cancel•No*** aborts the command.

LOADING System Exclusive Data from Disk and SENDING to an External Device

- Connect the MIDI Out of the ASR–10 to the MIDI In of the receiving device.
- Press ***Load***, then ***System•MIDI***. Scroll until the desired Sys-Ex file is selected. The display looks something like this:



- Press ***Enter•Yes***. The display reads LOADING <filename>.
- The display then reads CANCEL=QUIT ENTER=SEND.
- Press ***Enter•Yes*** to send the Sys-Ex file to a receiving device, or
- Press ***Cancel•No*** to abort the command.

After you have sent your MIDI Sys-Ex file to a receiving device, the CANCEL=QUIT ENTER=SEND screen returns. You can retransmit the data by pressing ***Enter•Yes***, in case it was not properly received the first time for some reason.

- Once the data has been successfully received by the external device, you can, if you wish, make a back-up copy of the file onto a different disk.
- Press ***Cancel•No***. The ASR–10 asks RESAVE FILE TO DISK?
- Press ***Enter•Yes*** to save the Sys-Ex file to disk, or ***Cancel•No*** if you don't want to make a backup.

CMD SYSTEM•MIDI	**WRITE DISK LABEL**
	Press Command / System•MIDI / scroll using the arrow buttons

This command is used to label ASR–10 disks with a seven character Disk I.D. These labels are used to give each of your ASR–10 disks a unique identity, thereby allowing banks to load from multiple disks.

To label an ASR–10 disk:

- Select WRITE DISK LABEL.
- Press ***Enter•Yes***.
  The display will read USE CURRENT LABEL? Pressing the ***Enter•Yes*** allows you to use the disk label from the last disk read by the drive. This can speed up the labelling process, as it is likely you'll want to keep the same first four alpha-numeric characters, changing only the three-digit number at the end of the label. If not, pressing ***Cancel•No*** defaults to DISK000.
- Use the data entry controls to name the disk.
- Press ***Enter•Yes***. The display briefly reads DISK COMMAND COMPLETED, then returns to the Command/System•MIDI page.

CMD SYSTEM•MIDI	COPY SCSI DRIVE
	Press Command / System•MIDI / scroll using the arrow buttons

This command enables you to copy the data from one SCSI Storage Device to another of the same capacity (they must have exactly the same capacity for the process to work) or copy data from one cartridge of a removable SCSI Storage Device to another cartridge. This command does *not* de-fragment the files as it copies them from the source drive to the destination drive. It merely copies the data sector by sector.

SCSI Drive Hook-Up

When using the COPY SCSI DRIVE command to copy data between two drives of the same size, connect a SCSI cable between one port of one drive to the SCSI connector on the rear of the ASR–10. Then, run a second SCSI cable from the first drive's remaining SCSI port to a port on the second drive. The order of the drives does not matter. However, each drive must have a different SCSI ID number in order for the ASR–10 to differentiate between the source and destination drives. Also, it is time to stress again that the two drives must be the same size (in memory capacity) for the process to work properly. The easiest way to ensure this is to use the same model by the same manufacturer for both drives. However, several manufacturers use the same drive mechanism in their drives, which will also work. Call the drive manufacturer to get complete details on your specific drive(s).

Note: If you're hooking up multiple drives, refer to the *Small Computer System Interface (SCSI) Manual* for instructions on how to correctly set termination for each individual drive.

For copying data between two cartridges of a removable drive, connect a SCSI cable between one of the SCSI ports on the drive and the SCSI connector on the ASR–10.

Using the COPY SCSI DRIVE Command

The following procedure is the basic method for copying data between two SCSI Storage Devices of the same size or two cartridges of a removable drive.

- With the drive(s) connected as stated above, press ***Command***, followed by ***System•MIDI***. This places you on the Command/System•MIDI page.
- Use the data entry controls to scroll to the COPY SCSI DRIVE command. Press ***Enter•Yes***.
- The display reads FROM DRIVE=SCSI #. Use the ***Up/Down Arrow*** buttons to set the SCSI ID number to that of the source drive (the drive that contains the data you wish to copy). For more information on SCSI ID number, refer to the *Small Computer System Interface (SCSI) Manual* that you received when SCSI was installed on your ASR–10. Press ***Enter•Yes***.
- The display now reads TO DRIVE=SCSI #. Use the ***Up/Down Arrow*** buttons to set the SCSI ID number to that of the destination drive (the drive to which you wish to copy the data). If copying between two cartridges of a removable drive, set the TO DRIVE and FROM DRIVE parameters to the same number. Press ***Enter•Yes***.
- If the destination drive needs to be formatted, the ASR–10 will display the message FORMAT SCSI DRIVE. Pressing ***Enter•Yes*** will cause the drive to be formatted. Immediately afterward, the ASR–10 begins copying the data from the source drive to the destination drive.

Note: When using a removable drive, the ASR–10 will issue prompts for inserting the source and destination disks. These refer to the separate cartridges, and are switched in the same manner as floppy disks in the COPY FLOPPY command. It is also very important to let the drive "spin-up" each time you switch between Source and Destination cartridges *before* hitting the ***Enter•Yes*** button to continue.

CMD SYSTEM•MIDI	**BACKUP/RESTORE**
	Press Command / System•MIDI / scroll using the arrow buttons

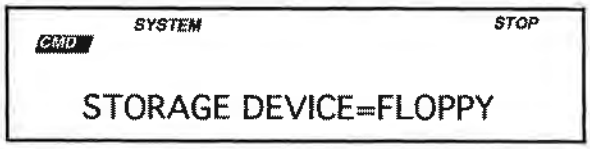
The BACKUP/RESTORE command allows you to make a backup copy of all of the information on a SCSI Storage Device. If the drive later fails and must be reformatted, or if you are switching to another drive, you can restore the drive from the backup copy.

- Press ***Command***, then ***System•MIDI***, and scroll until "BACKUP/RESTORE" appears in the display.
- Press ***Enter•Yes***. The ASR–10 will show:

This allows you to select either the BACKUP or RESTORE function. Use the ***Up/Down Arrow*** buttons or the ***Data Entry Slider*** to select the correct function.

- Press ***Enter•Yes***, and the ASR–10 will display:

This allows you to select the device that you want to backup or restore. You may only select SCSI 0 through 7. Press ***Enter•Yes***. The ASR–10 will display:



This allows you to select the device that you want to use to store the backup data. You may select SCSI 0 through 7, or the FLOPPY drive. Press ***Enter•Yes***, and the ASR–10 will proceed with either BACKUP or RESTORE.

BACKUP

Note: If you are using removable cartridges as your backup storage device, you must boot the ASR–10 with a formatted cartridge in the drive. Otherwise, the ASR–10 will not recognize the drive as a valid device. Subsequent cartridges need not be formatted beforehand; they will be automatically formatted as part of the backup procedure.

The BACKUP utility first scans the SCSI Storage Device to see how many disks (or cartridges) will be necessary to hold the backup data. While this scan occurs, you will see the names of directories appear briefly in the display. When the scan is finished, the ASR–10 displays:

YOU NEED nn DISKS, OK? ("nn" is the number of disks needed to hold the backup data).

- If you press ***Cancel•No***, the command will abort without copying any data.
- If you have enough disks (or cartridges), press ***Enter•Yes***.

If the SCSI Storage Device contains an ASR–10 Operating System, the ASR–10 will first copy the operating system onto the backup storage device. Next, the ASR–10 saves a file that contains a "snapshot" of the directory structure for the SCSI Storage Device. This file is named BACKUP FILE, and appears as a SYSTEM EXCLUSIVE file; however, it contains no valid Sys-Ex data. After these two files, the ASR–10 proceeds to copy each file on the SCSI Storage Device. Whenever a new disk (or cartridge) is necessary, the ASR–10 will prompt:

Insert a disk and press ***Enter•Yes***. If you insert an unformatted disk (or cartridge), the ASR–10 will automatically format it. If the disk is not blank, the ASR–10 will erase the files on the disk before proceeding to copy files to it. As you insert each disk, be sure to write the disk number somewhere on the disk label. If the ASR–10 encounters errors reading from or writing to the SCSI Storage Device, it will display a descriptive error message and wait for you to press ***Enter•Yes***. After you press ***Enter•Yes***, the ASR–10 will display CONTINUE ?

If you press ***Enter•Yes***, the ASR–10 will move on to the next file and proceed with the backup. When all of the files have been copied, the ASR–10 will go back to the Command page, displaying BACKUP/RESTORE. The actual backup may take more or less disks than the original estimate, depending upon errors reading files and bad sectors encountered writing files.

RESTORE

The RESTORE utility first checks the SCSI Storage Device to see if it needs formatting or erasing. If so, it will automatically format and erase the disk. Next, the ASR–10 displays:

After you insert the first backup disk (or cartridge) and press ***Enter•Yes***, the ASR–10 copies the OS (if necessary) from the backup disk. Then, it loads the BACKUP FILE and recreates all of the directories on the SCSI Storage Device. After this is completed, if you are restoring to a different SCSI device number than the one you backed up, the ASR–10 will display TRANSLATE SCSI x TO y ?

Where x is the SCSI ID of the storage device you backed up, and y is the SCSI ID of the storage device you are restoring. Pressing the ***Enter•Yes*** button here will cause the ASR–10 to change all BANKS and MACRO files that contained file paths for x to file paths which point at y. Pressing ***Cancel•No*** here will not abort the command; it merely answers "no" to the question.

Note: If you restore to a different device than the one you backed-up, and you answer "no" to the translate question, most of your banks will probably be useless.

The ASR–10 will restore the files one at a time. It will prompt for you to insert each disk when it is necessary. If you insert the wrong-numbered disk, the ASR–10 will respond WRONG DISK INSERTED.

If this happens, insert the correct disk and press ***Enter•Yes***.

Notes About BACKUP/RESTORE

The RESTORE utility will automatically defragment the SCSI Storage Device, resulting in faster access times for some files.

If you have backed-up your SCSI Storage Device and you accidentally erase a file, you don't need to restore the entire drive to get back that one file. Any of the copied files may be loaded directly from the backup disks without going through the restore process.

If you are backing up to removables, and it is your experience that they are faster when formatted with an INTERLEAVE other than 1, pre-format the necessary cartridges using the FORMAT command that allows you to specify the INTERLEAVE.

CMD SYSTEM•MIDI	**FORMAT SCSI DRIVE**
	Press Command / System•MIDI / scroll using the arrow buttons

This command is for formatting a SCSI Storage Device for use with the optional SP-3 Small Computer System Interface. See the *Small Computer System Interface (SCSI) Manual* for details.

Macros

What Is a Macro?

It may have occurred to you that with the large number of files that a SCSI Storage Device can contain, it would be nice to be able to get directly to a directory or specific file quickly rather than having to scroll through all your files. Macros allow you to do this. In the ASR–10, a macro is a shortcut command that allows you to get to a file or directory on your SCSI Storage Device (or floppy) with just two or three button presses.

Complete understanding and utilization of macros is very important to the effective use of your SCSI Storage Device. Spend time to make sure that these concepts are clear to you, and that you set up macros to get you to all the directories and files that you use often.

Organizing your files in directories makes it easier to locate an individual file, but moving up and down through directories can become tedious if you are working with a specific set of files that are in different directories. Macros allow you to assign a number (1-28) to an individual file location. You use or invoke the macro by holding down ***Load*** while entering the macro number (using the numeric keypad), and when you release ***Load***, the ASR–10 will automatically go to the correct directory for the requested file and display its name.

The group of up to 28 currently defined macros may be stored as a *Macro File*. These macro files appear along with directories on the Load/System•MIDI page. You can have many macro files on disk, but the only macros that are in effect are the macros from the most recently loaded macro file. Only one can be loaded in at a time, giving you access to twenty-eight macros. The best place to save these files is in the root directory. The default macro file created by the formatting procedure is located in the Root Directory of the SCSI Storage Device, and is called MACROFILE 1. A macro file having this or any other name will be automatically loaded at bootup if it is found as file number 5 in the Root Directory. If you want to use other macros, the first thing you should do is load the desired macro file after you boot from the SCSI Storage Device.

The macro file created by the formatting procedure, named MACROFILE 1, contains macros predefined for getting to the various directories in the default directory structure, as follows:

1 FLOPPY DRIVE	**2** FACTORY SNDS	**3** MY SOUNDS
	5 FACTORY SEQS	**6** MY SEQUENCES
	8 FACTORY BNKS	**9** MY BANKS
	0 MACRO FILE	

Note: All macro operations are performed with the ***Load*** button held down, and take effect when the ***Load*** button is released.

Loading a Macro File

- Press ***Load*** then ***System•MIDI***, and scroll to the macro file that you wish to load.
- Press ***Enter•Yes*** to load a new set of 28 macros into the ASR–10.

Creating a Macro

- Find the file or directory to which you want to assign a macro. Press and hold the ***Load*** button, type any number from 1 to 28, while still holding ***Load***, press ***Enter•Yes***, then release ***Load***. The macro assignment will remain in effect until another macro file is loaded or the ASR–10 is rebooted.

Invoking a Macro

- Press and hold the ***Load*** button, type and release the number of the desired macro. When you release the ***Load*** button, the ASR–10 will go to the file that is assigned to the macro. If the file is a directory, you will need to press ***Enter•Yes*** to actually enter the directory. If no file is assigned to the macro, the macro number will be displayed, and nothing else will happen. In MULTI and MONO B modes, incoming MIDI program changes 101-128 will invoke Macros 0-27.

Saving a Macro File

- First, invoke macro 0 (zero) to get back to the proper directory for the currently loaded macro file (macro 0 is a special macro that is always automatically assigned to the currently loaded macro file and cannot be reassigned. It is used to allow you to easily get back to the directory where the macro file that you last loaded is located. This makes saving your modified macro files easier).

- Press ***Command***, then ***System•MIDI***, and scroll to the SAVE MACRO FILE command and press ***Enter•Yes***. The name of the current macro file will be displayed.
- The display will show FILE NAME = MACROFILE 1. If you want to create a new macro file, edit the name of the file, and press ***Enter•Yes*** when you are ready. The new macro file will be saved in the same directory as the old macro file(s).
- If you wish to update the old file, just press ***Enter•Yes***. When the ASR–10 displays DELETE OLD VERSION? press ***Enter•Yes*** only if you wish to replace the old macro file with the new set of macros. This will save your newly defined set of macros into the macro file that was last loaded.

You can have as many different macro files as you want by giving them different names. You may also save them into any directory, although it is a good idea to save them in the root directory so that they are easy to load when you boot up. Remember that macros can take you to any file or directory, no matter how far down the file path. Also remember that you must resave the macro file any time you add new macros and want them to be there the next time you boot the ASR–10.

Hint: A good example of the use of macros would be to assign macro 11 to a directory consisting solely of bank files. Now, whenever you want to save your current group of instruments (and any saved song file) as a bank, simply invoke macro 11 that selects your bank directory, and then use the SAVE BANK command (press ***Command***, then ***Instrument***, and scroll to SAVE BANK). This saves you from having to locate your bank directory each time. Another useful application of macros is the transfer of files from floppy disks onto your SCSI Storage Device.

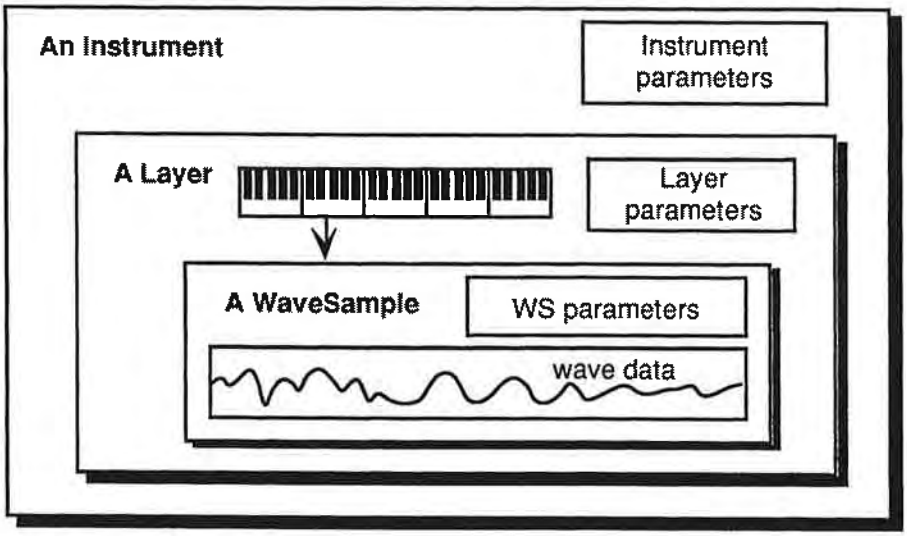
Instrument, Bank, & Preset Concepts

Section 3 — Instrument, Bank, and Preset Concepts

This section offers a basic overview of the concepts involved in selecting, creating, editing, and understanding Instruments, Banks and Performance Presets. For more detailed descriptions of the actual parameters, refer to the following section.

Instruments

The ASR–10 keeps all its sounds in a format called an *Instrument*. This format is used for all disk storage. The internal structure of an Instrument has three levels: Instrument, Layer, and WaveSample. Up to 127 WaveSamples are grouped together into layers, and there can be up to eight layers in the instrument. Their relationship is shown in the following diagram:



At the instrument level, there are parameters that affect the whole instrument: for example the MIDI functions and the patch select settings. These parameters are found on the Edit/Instrument page. Parameters that affect each layer are found on the Edit/Layer page. Each distinct area of WaveSample editing (i.e. Pitch, Filter, Amp) has its own Edit page.

Loading an Instrument

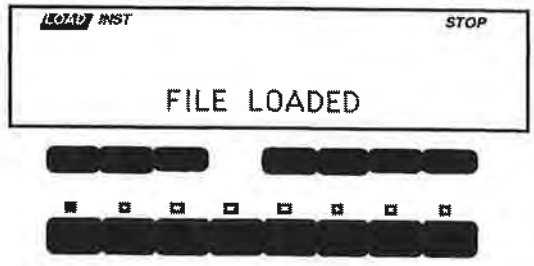
You can load up to eight different instruments into the ASR–10 at once (within the limits of memory). First, insert a disk containing one or more instrument files into the disk drive.

- Press ***Load.*** The LOAD indicator flashes.
- Press ***Instrument.*** The display looks like this:

When the LOAD indicator is flashing, the ASR–10 is showing you disk files (think of it as a question mark — the ASR–10 is saying "Load the file showing on the display?"). Pressing the ***Up/Down Arrow*** buttons takes you through the files on the disk. If there are none, the display will read "NO INST OR BANK FILES."

Whenever a disk file is displayed as above, you can press the ***Left*** or ***Right Arrow*** button to see the size of that file in *Blocks* (a Block is 256 samples; 4 Blocks=1K sample words). Press the ***Left*** or ***Right Arrow*** button again to return to the file name.

- Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to view the various instrument files on the disk. Each file has its own File Number. When an instrument file is showing, the INST indicator is lit. The BANK Indicator will light when a bank file is showing.
- Find the instrument you want to load, and press ***Enter•Yes***. The display will say PICK INSTRUMENT BUTTON. The ASR–10 is asking in which instrument location you want to load the sound into.
- Press any of the eight ***Instrument•Sequence Track*** buttons. The ASR–10 will begin immediately loading the instrument into the selected location. The display reads LOADING FILE... and the LED *flashes* red while the instrument is being loaded.



Once the instrument has been loaded, the display briefly shows "FILE LOADED." The LED above the ***Instrument•Sequence Track*** button stops flashing and remains solidly lit red, indicating that there is now an instrument loaded in that location which can be selected by pressing that button.

If you tell the ASR–10 to load an instrument into a location that already has an instrument loaded (red LED), the new instrument will be loaded into that location and the one that was there will be automatically deleted.

Saving an Instrument to Disk

After you have created a new ASR–10 instrument, or made changes to an existing instrument, you can save the instrument to a formatted ASR–10 disk with the following steps:

- Insert a formatted disk into the drive. If you don't have a formatted disk, format one before proceeding.
- Press ***Command***, then ***Instrument***. This puts you on the Command/Instrument page, that contains all the instrument-related commands. Check to make sure that the instrument you want to save is selected. If not, press its ***Instrument•Sequence Track*** button to select it.
- Press the ***Left*** or ***Right Arrow*** button until the display reads SAVE INSTRUMENT (or use this shortcut — after pressing ***Command***, double-click on the ***Instrument*** button).
- Press ***Enter•Yes***.
- Edit the Instrument Name (if needed):

The display shows the current name with a cursor (underline) beneath the first character. If you want to give the instrument a new name, do so at this time. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the underlined character, then press the ***Left*** or ***Right Arrow*** button to move the underline. Repeat until the display shows the name you want (if you don't need to rename the instrument, just skip this step).

- Press ***Enter•Yes***.
  The display will read SAVING <INST NAME> while the instrument is being saved. Note that you can continue playing the keyboard while the instrument is being saved, and you can select other instruments. You cannot get out of Command mode until the save procedure is done.
- If there is already an instrument file with the same name on the disk, the display will ask DELETE OLD VERSION? Press ***Enter•Yes*** to save the instrument, replacing the one on the disk. This is for updating instruments to which you have made changes. Or press ***Cancel•No*** to abort the procedure.
- If there is not enough free space on the disk for the instrument, the display will say NOT ENOUGH DISK SPACE. Save the instrument to another disk (or delete some files from the disk).

Banks

Banks provide a way to load a whole group of instruments and sequences into the ASR–10 with a few button presses. When you save a Bank to disk, it is like taking a "snapshot" of the contents of the ASR–10 internal memory. The Bank file stores the following information:

- Which instruments are loaded into each ***Instrument•Sequence Track*** location
- Which song and its related sequences are loaded into the internal memory
- Up to eight discrete Performance Presets
- The Bank Effect and its parameter settings
- The Edit/Track MIX, PAN, OUT, and EFFECT MOD CONTROL setting for each ***Instrument•Sequence Track***

For example, you might have a piano loaded into ***Instrument•Sequence Track 1***, a bass in ***Instrument•Sequence Track 2***, and drums in ***Instrument•Sequence Track 3***, and you have created a number of performance presets containing different keyboard configurations of those instruments. Let's say you also have a song in memory, that is composed of 12 sequences. If you now save the contents of memory as a Bank, you can later call up this exact setup by loading the Bank.

Important Notes About Banks

- The Bank file on a disk doesn't contain the actual instrument and song files — it is just a set of instructions telling the ASR–10 what to load and where to load it. All the instruments and the song in a Bank must be saved to disk separately (or loaded in from a disk with a proper Disk Label ID) *prior* to saving the Bank.

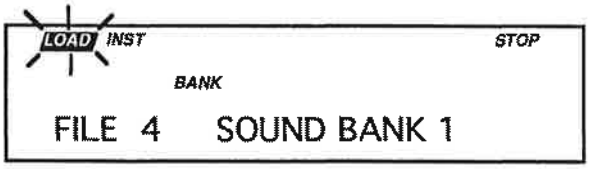
- Any instruments already loaded into locations not affected by the Bank will be left intact (memory permitting). You can select and play such instruments while the Bank loads. Also, any instruments included in the Bank that are already in memory will be left there and not reloaded, even if they are in different ***Instrument•Sequence Track*** locations in the new Bank. In such a case the ASR–10 will simply move the instrument to the new location.

- The instrument and song files referred to by a Bank can be on many different disks. Each time you load an Instrument or Song, the ASR–10 "remembers" what disk it was loaded from. When you load a Bank file, the ASR–10 will ask by name for any disks it needs to load the files included in the Bank. This is why it is very important that you give each of your disks a unique name (Disk Label ID) and write that name on the disk itself.

- If there was a song in memory when the Bank was saved, the ASR–10 will automatically load the song as part of the Bank. If you want the Bank to load without a song, erase the contents of the sequencer memory before saving the Bank to disk.

To Load a Bank

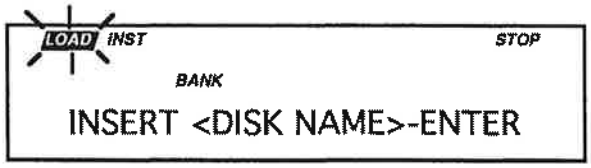
- Press ***Load***, then ***Instrument***.
- Use the ***Up/Down Arrow*** buttons to view the different files.



When a Bank file is showing, the BANK indicator lights on the display along with the INST indicator.

- Once a Bank file is showing, press ***Enter•Yes***. The ASR–10 will begin loading the instruments and the song data. As it loads each instrument the display tells you what it's doing. As soon as any of the instruments are finished loading, you can select that instrument and play while the rest of the Bank loads.

At some point in the loading process, you may encounter the following message:



This prompt is informing you that an instrument or song saved as part of the Bank is on a different disk than the disk currently in the drive. When this occurs:

- Eject the current disk in the disk drive and replace it with the disk that matches the Disk Label ID requested by the ASR–10 (you are writing these names on your disks, aren't you?).
- Press ***Enter•Yes***. The ASR–10 will resume loading until completed, or until the point when it needs another disk.
- When it has finished loading the instruments, the ASR–10 will load the song (if any) and then set up any copied instruments included in the Bank.

Saving the Contents of Memory as a Bank

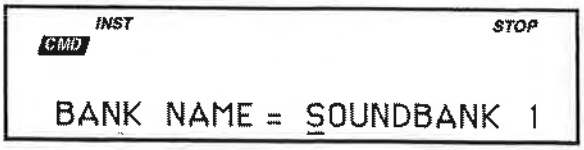
Saving a bank is like taking a "snapshot" of the contents of the ASR–10 memory. When you later load a bank, the ASR–10 "looks" at that snapshot and tries to recreate what was in memory when the bank was saved. You can use banks to automatically load a new group of instruments and/or a new song.

Banks are valuable for this reason: ASR–10 songs and sequences have no way of knowing which instruments are loaded into various ***Instrument•Sequence Track*** locations. Suppose, for example, you record a track with a piano instrument, but then load a kazoo sample into the instrument location where the piano was. The track will now play on the kazoo. By using banks you load instruments into their "proper" location. The Bank file stores the following information:

- Which instruments are loaded into each ***Instrument•Sequence Track*** location
- Which song and its related sequences are loaded into the internal memory
- Up to eight discrete Performance Presets
- The Bank Effect and its parameter settings
- The Edit/Track MIX, PAN, OUT, and EFFECT MOD CONTROL setting for each ***Instrument•Sequence Track***

It's easy to save the contents of the ASR–10 memory as a bank. Just remember that any instruments in the internal memory, as well as the song, must be saved separately before they can be saved as a bank.

- Save any instruments in memory to disk using the SAVE INSTRUMENT command on the Command/Instrument page.
- Save the song to disk (if you want a song as part of the bank) using the SAVE SONG + ALL SEQS command on the Command/Seq•Song page.
- Press ***Command***, then ***Instrument***.
- Press the ***Left*** or ***Right Arrow*** button until the display reads SAVE BANK.
- Press ***Enter•Yes***. The display shows:



The display shows the current name with a cursor (underline) beneath the first character. If you want to give the bank a new name, do so at this time. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the underlined character, then press the ***Left*** or ***Right Arrow*** button to move the underline. Repeat until the display shows the name you want. (If you are updating an existing bank, don't rename the bank; just press ***Enter•Yes*** and then press ***Enter•Yes*** once more in response to the question DELETE OLD VERSION?).

- Press ***Enter•Yes***.

Note: It's not necessary for the instruments and song to be saved on the same disk as the bank. Each instrument, the song, and bank can be saved on separate disks, but make sure the disks are labelled (WRITE DISK LABEL command found on Command/System•MIDI page) so that the bank can locate the information properly.

Performance Presets

Once you have a number of instruments loaded into the ASR–10, there are so many possible combinations for selecting, stacking, mixing, and manipulating the ranges of those instruments that it can get a bit hard to manage during a performance. That is where Performance Presets come in.

For each bank, you can create eight performance presets which allow you to store different keyboard configurations. By selecting a performance preset you can instantly recreate the state of the "pile" at the time the preset was created. These eight performance presets are saved with the bank, and are loaded each time a bank is loaded.

Within each preset, the following information is saved for instant recall:
- Which instruments are selected, deselected, or stacked (on the "pile" in Load mode)
- The current "locked in" status of the patch select buttons for each ***Instrument•Sequence Track***
- The Edit/Track MIX (volume level), PAN, OUT Bus routing, and EFFECT MOD CONTROL settings for each ***Instrument•Sequence Track***
- The Edit/Instrument MIDI OUT CHANNEL number, MIDI OUT PROGRAM number, PRESSURE mode setting, MIDI STATUS setting, PATCH SELECT MODE setting, INST (key) RANGE , and XPOS (transpose) settings for each ***Instrument•Sequence Track***

Selecting Performance Presets

When the ASR–10 is in Load mode (the LOAD indicator solidly lit), pressing the *Page* buttons numbered 1-8 (not the ***Instrument•Sequence Track*** buttons) will select the eight performance presets once they've been created.

To select Performance Presets when in Edit, Command, or (flashing) Load mode:
- *While holding down* ***Load***, press and release ***Instrument***.
- The display will show:

Press a number (page) button to select a Performance Preset. After pressing a number button, the display will return to (solid) Load mode.

If there are no Performance Presets defined, the display will briefly show the following message and then return to (solid) Load mode:

If a selected Performance Preset has not been defined, the display will briefly show the following message and then return to (solid) Load mode:

Note: When the LOAD indicator is flashing, the display is showing you disk files for loading. When the LOAD indicator is lit but not flashing, the display is showing you the name(s) of the instruments in the Internal Memory.

Important Notes about Performance Presets

- Pressing Page button #0 will deselect all the instruments in memory and return them to the original settings for INST (key) RANGE, XPOS (transpose), MIX, etc. that were in effect when they were loaded (or most recently saved to disk). This will undo the effects of any presets currently in effect. You can press Page button #0 at any time (whether a preset is in effect or not) to return all your instruments to their "starting points."
- If you save an instrument to disk while a preset is in effect, all the current values will be saved with the instrument. Be careful that you don't accidentally save an instrument with some strange values that you had meant to be part of a temporary setup. You can select Page button #0 before saving the instrument to avoid this.
- If no preset has been created for one of the eight possible locations, pressing that number will have no effect.

Creating a Performance Preset

Whenever you have set up some configuration of instruments (in Load mode) that you want to preserve for later recall, you can save that configuration as a performance preset with the following steps:

- In Load mode, set up the instruments in memory in some configuration you want to preserve.
- Press ***Command***, then ***Instrument***.
- Press the ***Left*** or ***Right Arrow*** button until the display shows CREATE PRESET.
- Press ***Enter•Yes***. The display shows:

In this example, preset number 1 is undefined (*UND* will only appear when a preset is undefined).

Hint: In (solid) Load mode, there is a quick way to get to this screen. First set up the Instruments in memory in the configuration that you want to preserve. While holding down the ***Command*** button, press and release the ***Instrument*** button.

- Use the ***Data Entry Slider*** or the ***Up*** and ***Down Arrow*** buttons to select a preset number 1-8. This will determine where (of the eight possible performance preset locations) the current keyboard configuration will be saved.
- Press ***Enter•Yes***. The display reads COMMAND COMPLETED and the preset is made.
- Press ***Load*** to return to the "Pile." Pressing the number that you selected for the preset will now instantly recall this keyboard configuration. Repeat this procedure for each performance preset you want to create, selecting a new number (from 1-8) for each.

Note: Performance presets are saved to disk only when you save the contents of memory as a bank. When you load a bank, any presets that were present at the time the bank was saved will be recreated.

Suggested Uses for Performance Presets

The most obvious use for performance presets is to have instant access to different combinations of instruments selected, stacked, and mixed in various ways. Here are some other possibilities:

- Easier access to patch select variations. Suppose you have selected the piano sound and "locked in" some patch variation (by holding down the desired combination of patch select buttons, and then pressing an ***Instrument•Sequence Track*** button). By creating a preset while the alternate patch is "locked in," you can later return to that sound just by selecting the preset.
- Controlling remote MIDI devices. Suppose you have created one or more MIDI instruments for sequencing remote MIDI devices from the ASR–10. Each performance preset can store a different MIDI OUT PROGRAM number, VOLUME setting, INST (key) RANGE and XPOS (transpose) amount for each of those remote MIDI devices. When you select a preset, each instrument that has a status of MIDI or BOTH (SEND KEYS TO= on the Edit/Instrument page) will transmit a program change and MIDI volume message on its designated MIDI OUT CHANNEL (assuming TRANSMIT ON = INST CHAN on the Edit/System•MIDI page) to the appropriate remote device. Performance presets let you store up to eight totally different configurations. This allows you to easily recall stacks and combinations of remote MIDI devices.

Performance Tip — Creating Keyboard Splits

One of the most common uses of presets is a keyboard split, in which your left hand plays an "accompaniment" sound and your right hand plays a "solo" sound. This is easy to set up on the ASR–10 by stacking the two sounds and assigning them different INST (key) RANGES. You can save this arrangement as a preset so that any time you select that preset, you'll recall the keyboard split. In fact, using this method you can create up to eight split points across the keyboard.

Since each ***Instrument•Sequence Track*** in the keyboard split has its own XPOS (transpose) value, you are not limited to bass notes on the piano just because it's on the bottom of the keyboard. On the Edit/Instrument page, set the INST (key) RANGE to the keys you want to play and then transpose the sound up or down until it's playing in the range you want.

Instrument, Bank, & Preset Parameters

Section 4 — Instrument, Bank, and Preset Parameters

This section offers detailed descriptions of the commands and parameters used in selecting, creating, and editing Instruments, Banks and Performance Presets. For an overview of the concepts involved, refer to the previous section.

Edit/Instrument Page

The parameters on this page are used to modify the individual Instrument(s). To access these parameters press ***Edit***, then ***Instrument***.

Hint: When Edit/Instrument parameters are displayed, pressing a loaded ***Instrument•Sequence Track*** button will display the Edit/Instrument values for the selected Instrument. This feature provides quick access to the same parameter for different instruments when setting instrument parameter values relative to one another.

EDIT INST	**PATCH (Layer Enable/Disable)**
	Press Edit / Instrument / 0

Using the Patch Select Jack

When the optional SW–10 Dual Foot Switch is plugged into the Patch Select jack, it can be programmed to provide instant access to up to four completely different sounds within each instrument. The alternate sound might be a different set of samples, or the same samples with different envelopes, or a chorused version of the sound, or perhaps a different range of an instrument.

What you are doing when you press either or both of the patch select foot switches is selecting different layers to play. A layer is a group of WaveSamples spread across the keyboard — each instrument can have up to eight layers. For each of the four possible patches, any of these eight layers can be made to play or to remain silent.

- The patch select foot switches are "momentary" — an alternate patch will play only on notes played while the foot switch is held down unless you "lock in" a Patch.
- To "lock in" a particular patch so that it will play without having to hold any foot switches: while holding down the patch select foot switch(es) corresponding to the patch you want to "lock in," simply press the ***Instrument•Sequence Track*** button.
- To return to the default patch (neither patch select foot switch down) after you have "locked in" one of the other patches, press the ***Instrument•Sequence Track*** button again (with neither of the patch select foot switches held down).

Here's how a typical instrument might react when you press the different patch select foot switches:

With the Optional SW-10 connected to the Patch Select jack:

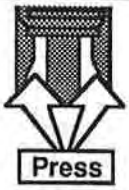
With no foot switches pressed: The Display shows: `00 PATCH= 1 2 3 4 - - - -`

With the right foot switch pressed:

The Display shows: `0* PATCH= 1 - - - - - - -`

With the left foot switch pressed:

The Display shows: `*0 PATCH= - 2 - - - - - -`

With both foot switches pressed: 

The Display shows: `** PATCH= 1 2 - - - - - -`

- A number means that a given layer is enabled, and will play in that patch;
- A dash means that the layer is disabled for that patch and will not play.

Programming Patches

The PATCH parameter allows you to program a layer—or combination of layers—to either play or not play depending on which patch select foot switches are pressed.

To program patches:

- Select the instrument on which you would like to program patches.
- Select the 00 PATCH parameter (no patch select foot switches pressed).
- With no patch select foot switches pressed, use the ***Left/Right Arrow*** buttons to select the individual layers and the ***Up/Down Arrow*** buttons to enable and disable them. This will program the default patch (the one that will play with no patch select foot switches pressed).
- Once the default patch is programmed, press the left patch select foot switch. Notice that the left hand character in the display is now an asterisk, rather than a zero. This indicates the status of the particular patch select foot switch.
- With the left patch select foot switch held down, use the ***Left/Right Arrow*** buttons to select the individual layers and the ***Up/Down Arrow*** buttons to enable and disable them, as you did with the default patch.
- Repeat the procedure with the right patch select foot switch held down, then with both patch select foot switches held down.
- Remember to resave the instrument after you've made any changes.

The Patch Select foot switches are transmitted and received over MIDI as MIDI controller #70.

EDIT INST	KEYDOWN LAYERS
	Press Edit / Instrument / 1

This parameter determines which layer(s) will be triggered when a MIDI keydown is received.

- Select KEYDWN LAYERS=########. Use the cursor and the ***Up/Down Arrow*** buttons to select either a number (layer enabled on keydown) or a dash (layer disabled on keydown) for each layer.

EDIT INST	KEYUP LAYERS
	Press Edit / Instrument / 2

This parameter determines which layer(s) will be triggered when a MIDI keyup message is received. A good example of this would be the "thunk" sound made by a harpsichord when you release a key. This release sound can be sampled separately and placed in a layer that you designate as a keyup layer. If you want a layer to be triggered *only* when keys are released, be sure to turn that layer off with the KEYDWN LAYERS parameter.

- Select KEYUP LAYERS=########. Use the cursor and the ***Up/Down Arrow*** buttons to select either a number (layer enabled on keyup) or a dash (layer disabled on keyup) for each layer.

EDIT INST	MIDI OUT CHANNEL
	Press Edit / Instrument / 3

Use this parameter to determine which MIDI channel sequencer playback and Smart Transmit™ will transmit on, *if* the TRANSMIT ON= parameter on the Edit/System•MIDI page is set to INST CHANNEL. Any of the 16 MIDI channels can be selected.

If TRANSMIT ON=BASE CHANNEL, the ASR–10 will only Smart Transmit on the Base MIDI channel (also on the Edit/System•MIDI page) no matter which MIDI channel is selected.

Sequence playback will always transmit on the MIDI OUT CHANNEL, even if Edit/System•MIDI, TRANSMIT ON=BASE CHAN.

Also, if the instrument has been assigned LOCAL status, it won't matter what MIDI channel you assign here — playing that sequence will not transmit any MIDI information, and Smart Transmit will not work.

Range: MIDI channels 1 to 16

EDIT INST	**MIDI OUT PROGRAM**
	Press Edit / Instrument / 4

This parameter allows you to choose which program change number will be transmitted via MIDI when the instrument is selected. If the instrument's Edit/Instrument MIDI STATUS is set to LOCAL, it won't matter what this parameter is set to because the instrument will not transmit program changes. The MIDI OUT PROGRAM for each instrument is stored in the eight performance presets, so you can use presets to transmit new program changes from any instrument that has its MIDI STATUS set to BOTH, MIDI or EXT. Editing this value will transmit the program change number that is displayed.

Range: 1 to 128

Hint: MIDI OUT PROGRAM numbers can be entered directly from the numeric keypad from any mode. Here's how:

- While holding down the appropriate ***Instrument•Sequence Track*** button (the one that will transmit on the desired MIDI channel), use the numeric keypad (page buttons) to type in the program number that you want to transmit.
- Release the ***Instrument•Sequence Track*** button when you want to transmit the program change that you have entered. The program change will be transmitted *at the moment the button is released.*

EDIT INST	**PRESSURE MODE**
	Press Edit / Instrument / 5

If TRANSMIT=INST CHAN (on the Edit/System•MIDI page) then this parameter determines which type of pressure—channel or Poly-Key™, or no pressure at all—will be transmitted on the instrument's MIDI OUT CHANNEL. If the instrument MIDI STATUS= LOCAL or BOTH, this parameter determines what type of pressure (if any) the instrument will respond to when you play the keyboard.

When PRESSURE MODE=OFF:

- Local voices played from the MIDI keyboard or the sequencer will not respond to pressure.
- The sequencer will not record pressure into any tracks you record.
- The sequencer will not play back any key or channel pressure messages recorded on the track.
- The instrument will not transmit or receive pressure of either type via MIDI.

Note: You should select PRESSURE MODE=OFF whenever you don't want or need pressure for a given instrument, particularly when sequencing. Pressure data eats up sequencer memory rapidly, so whenever you record a track (whether it plays LOCALLY, via MIDI or BOTH), with a sound that doesn't respond to pressure, set this parameter to OFF. This will avoid wasting valuable sequencer memory, and will avoid clogging up the MIDI data stream with a lot of unnecessary pressure information.

When PRESSURE MODE=KEY:
- Local voices played from the MIDI keyboard will respond to key pressure only.
- The sequencer will record key pressure into any tracks you record.
- The sequencer will play back any key pressure messages recorded on the track, and will ignore any channel pressure on the track.
- The ASR–10 will only transmit key pressure out via MIDI; however,
- Either channel pressure or key pressure will be received via MIDI.

When PRESSURE MODE=CHAN:
- Local voices played from the MIDI keyboard will respond to channel pressure only.
- The sequencer will record channel pressure into any tracks you record.
- The sequencer will play back any channel pressure messages recorded on the track, and will ignore any key pressure on the track.
- The ASR–10 will only transmit channel pressure out via MIDI, however.
- Either channel pressure or key pressure will be received via MIDI.

Note: When SEQ REC SOURCE = MULTI on the Edit/Seq•Song page, the ASR–10 sequencer will always record both channel and key pressure (received via MIDI), no matter what the PRESSURE MODE parameter is set to.

At present, channel pressure is recognized by more MIDI devices than key pressure. If you are playing or sequencing an external MIDI device from the ASR–10 and pressure doesn't seem to have an effect, it could be the ASR–10 is set to transmit key pressure and the receiving instrument only recognizes channel pressure. In this case, set this parameter to PRESSURE MODE= CHAN when playing or sequencing that instrument.

EDIT INST	**MIDI STATUS**
	Press Edit / Instrument / 6

This parameter will determine whether the instrument will play LOCALLY (only on the ASR–10), or via MIDI, or BOTH. There are four possible settings:

- BOTH — The instrument will be played locally by the sequencer, and sequencer playback will transmit via MIDI when Edit/System•MIDI, TRANSMIT ON=INST CHAN. In addition, incoming MIDI will play internal voices.
- LOCAL. The sequencer will only play local (internal) voices, and will not transmit via MIDI. Incoming MIDI will also play local voices.
- MIDI — Sequencer playback will be transmitted via MIDI when Edit/System•MIDI, TRANSMIT ON=INST CHAN. The sequencer will not play ASR–10 voices at all. However, *incoming* MIDI will play internal voices. Use this status when you want to create MIDI-only instruments for sequencing Remote MIDI Devices from the ASR–10.
- EXT — This mode is similar to MIDI, except that incoming MIDI will *not* play internal voices.

Note: These settings are in effect only when TRANSMIT ON=INST CHANNEL on the Edit/System•MIDI page.

TRACK STATUS =	BOTH	LOCAL	MIDI	*EXT*
Playing the sequencer track (pressing ***Play***) plays local (internal) voices	Yes	Yes	No	No
Playing the sequencer track (pressing ***Play***) transmits via MIDI	Yes	No	Yes	Yes
Incoming MIDI data plays local voices	Yes	Yes	Yes	No

EDIT INST	**SIZE (in blocks)**
	Press Edit / Instrument / 7

This display shows how much internal memory the instrument occupies. This is a *readout only* and can't be modified. Size is shown in *Blocks*. One Block=256 samples (or words). Four Blocks=1k sample/word.

When you want to know if there's enough room to save a particular instrument to a particular disk, you can compare the instrument size (shown here), to the FREE DISK BLOCKS shown on the Edit/System•MIDI page.

EDIT INST	**INST NAME**
	Press Edit / Instrument / 8

You can use this parameter to change the name of an existing instrument, or name a new one that you've just created. The display shows the twelve-character name of the instrument with a cursor beneath the first character. Newly sampled or created instruments are given the default name UNNAMED INST.

- Select NAME=####. Use a combination of the ***Arrow*** buttons and ***Data Entry Slider*** to name/rename the instrument.

EDIT INST	**PATCH SELECT MODE**
	Press Edit / Instrument / 9

This parameter differs from the PATCH parameter (earlier in this section) in that this parameter allows you to permanently "lock in" one of the four patch variations, or to select a special HELD mode. The PATCH parameter, on the other hand, programs which combinations of layers will play for each of the four patch select assignments.

Setting	Patch used by track
LIVE	uses the current state to select a patch
-00-	always uses the 00 patch
-0*-	always uses the 0* patch
-*0-	always uses the *0 patch
-**-	always uses the ** patch
HELD	uses the patch set by the first key hit while pressing the patch select foot switch(es)

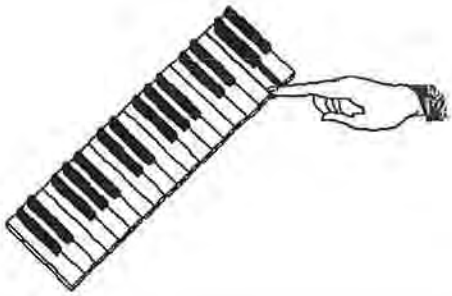
The HELD setting is special because it lets you easily hold a patch select variation by playing a key on the controller keyboard while pressing the patch select foot switch(es). The first key you play after pressing the patch select foot switch(es) will hold that patch select variation as the current patch for the track. You may then continue to play without having to press the patch select foot switch(es). If you wish to return to the 00 patch, press either of the patch select foot switches and release it *before* you play the next note. This will hold the 00 patch.

EDIT INST	**INST (Key) RANGE**
	Press Edit / Instrument / scroll using the arrow buttons

This parameter allows you to set the keyboard range of the instrument. Keyboard ranges are very important in the creation of keyboard splits and layers. The ASR–10 allows you to set the keyboard range in a very easy fashion.

- Select the LO=__ half of Instrument Key Range parameter, then:

1. Play Low Key on the MIDI Keyboard



2. Play High Key on the MIDI keyboard

Make sure after you've set the ranges that you change screens before playing the keyboard again. The Instrument key range parameter is always active when shown on the display. You can also use the ***Data Entry Slider*** and the ***Up/Down Arrow*** buttons to set the key range.

Range: KY0 to K11, C0 to G9

Note: Keys in the octave below C0 are displayed as KY0 through K11. These values are only accessable when the instrument is transposed up.

Note: If you set the Low Key *higher* than the High Key, the instrument is in effect turned off and will make no sound. If you do this, simply reset the parameters.

EDIT INST	**XPOS OCT - SEMI (Transpose)**
	Press Edit / Instrument / scroll using the arrow buttons

Each instrument can have its pitch transposed (raised or lowered) by octaves and semitones within an eight octave range. The transpose setting affects both the pitch played by Local ASR–10 voices and the key number transmitted via MIDI.

The ***Left/Right Arrow*** buttons will toggle the underline (showing the active field) between octave and semitone.

Octave Range: -4 to +4 octaves
Semitone Range: -11 to +11 semitones

Hint: It is possible to extend the transposition outside of the normal range by using the ***Up/Down Arrow*** buttons to edit the semitone value to a maximum of + or - 9 octaves and 11 semitones, although this may cause unexpected results.

Command/Instrument Page

CMD INST	CREATE NEW INSTRUMENT
	Press Command / Instrument / 0

Use this command to create a new instrument. This is useful for creating instruments that will only play out via MIDI (MIDI Instruments).

- Select CREATE NEW INSTRUMENT.
- Press ***Enter•Yes***. The display reads SELECT UNUSED INSTR=##.
- Press one of the unused ***Instrument•Sequence Track*** buttons.
- Press ***Enter•Yes***. The display reads COMMAND COMPLETED.

CMD INST	COPY INSTRUMENT
	Press Command / Instrument / 1

Use this command to copy the selected instrument. This is useful for making copies of an instrument in several tracks for sequencing. The copies don't use any internal memory, they just "point to" the data in the original instrument.

- Select COPY INSTRUMENT.
- Press ***Enter•Yes***. The display reads SELECT UNUSED INST=##.
- Press one of the unused ***Instrument•Track*** buttons.
- Press ***Enter•Yes***. The display reads COMMAND COMPLETED.

Shortcut:

- Make sure the instrument you want to copy is selected.
- While holding down the empty ***Instrument•Track*** button that you want to copy the *selected* instrument to, press ***Enter•Yes***.

CMD INST	DELETE INSTRUMENT
	Press Command / Instrument / 2

Use this command to delete the selected instrument from the Internal Memory. Note that this command has no effect on disk files, only on the Internal RAM Memory.

- Select DELETE INSTRUMENT.
- Press ***Enter•Yes***. The display reads DELETE <name of Instrument>?
- Press ***Enter•Yes***. The display reads COMMAND COMPLETED.

Shortcut:

- Make sure the instrument you want to delete is selected.
- While holding down the ***Instrument•Track*** button of the instrument you want to delete, press ***Cancel•No***.

CMD INST	SAVE INSTRUMENT
	Press Command / Instrument / 3

Use this command to save the selected instrument to floppy disk (or to a SCSI device). This is important because the ASR–10 does not save any Instruments, Banks, Presets or Sequence/Songs internally — if you were to power down, you would loose any unsaved information.

- Insert a formatted disk into the drive. If you don't have a formatted disk, format one before proceeding.
- Check to make sure that the instrument you want to save is selected. If not, press its ***Instrument•Sequence Track*** button to select it.
- Press the ***Left*** or ***Right Arrow*** button until the display reads SAVE INSTRUMENT (or use this shortcut — after pressing ***Command***, double-click on the ***Instrument*** button).
- Press ***Enter•Yes***.
- The display shows the current name with a cursor (underline) beneath the first character. If you want to give the instrument a new name, do so at this time. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the underlined character, then press the ***Left*** or ***Right Arrow*** button to move the cursor. Repeat until the display shows the name you want (if you don't need to rename the instrument, just skip this step).
- Press ***Enter•Yes***.
  The display will flash SAVING <INST NAME> while the instrument is being saved. Note that you can continue playing the keyboard while the instrument is being saved, and you can select other instruments. You cannot get out of Command mode until the save procedure is done.
- If there is already an instrument file with the same name on the disk, the display will ask DELETE OLD VERSION? Press ***Enter•Yes*** to save the instrument, replacing the one on the disk. This is for updating instruments to which you have made changes. Or press ***Cancel•No*** to abort the procedure.
- If there is not enough free space on the disk for the instrument, the display will say NOT ENOUGH DISK SPACE. Save the instrument to another disk (or delete some files from the disk).

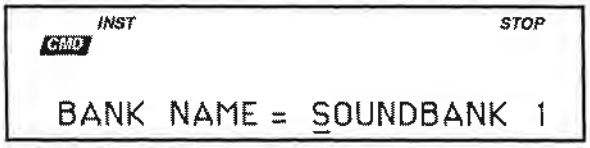
CMD INST	**SAVE BANK**
	Press Command / Instrument / 4

Use this command to save the current Bank. Banks provide a way to load a whole group of instruments and sequences into the ASR–10 with a few button presses. When you save a Bank to disk, it is like taking a "snapshot" of the contents of the ASR–10 internal memory. The Bank file stores the following information:

- Which instruments are loaded into each ***Instrument•Sequence Track*** location
- Which song and its related sequences are loaded into the internal memory
- Up to eight discrete Performance Presets
- The Bank Effect and its parameter settings
- The Edit/Track MIX, PAN, OUT, and EFFECT MOD CONTROL setting for each ***Instrument•Sequence Track***

It's easy to save the contents of the ASR–10 memory as a bank. Just remember that any instruments in the internal memory, as well as the song, must be saved separately before they can be saved as a bank.

- Save any instruments in memory to disk using the SAVE INSTRUMENT command.
- Save the song to disk (if you want a song as part of the bank) using the SAVE SONG + ALL SEQS command on the Command/Seq•Song page.
- Select SAVE BANK.
- Press ***Enter•Yes.*** The display shows:



The display shows the current name with a cursor (underline) beneath the first character. If you want to give the bank a new name, do so at this time. Use the ***Data Entry Controls*** to name the bank (as explained earlier in SAVE INSTRUMENT). (If you are updating an existing bank, don't rename the bank; just press ***Enter•Yes*** and then press ***Enter•Yes*** once more in response to the question DELETE OLD VERSION?).

- Press ***Enter•Yes.***

Note: It's not necessary for the instruments and song to be saved on the same disk as the bank. Each instrument, the song, and bank can be saved on separate disks, but make sure the disks are labelled (WRITE DISK LABEL command found on Command/System•MIDI page) so that the bank can locate the information properly.

CMD INST	CREATE PRESET
	Press Command / Instrument / 5

This command is used to create each of the eight Performance Presets within a Bank.

- In Load mode, set up the instruments in memory in some configuration you want to preserve.
- Press ***Command***, then ***Instrument***.
- Press the ***Left*** or ***Right Arrow*** button until the display shows CREATE PRESET.
- Press ***Enter•Yes***. The display shows:

In this example, preset number 1 is undefined (*UND* will only appear when a preset is undefined).

Hint: In (solid) Load mode, there is a quick way to get to this screen. First set up the Instruments in memory in the configuration that you want to preserve. While holding down the ***Command*** button, press and release the ***Instrument*** button.

- Use the ***Data Entry Slider*** or the ***Up*** and ***Down Arrow*** buttons to select a preset number 1-8. This will determine where (of the eight possible performance preset locations) the current keyboard configuration will be saved.
- Press ***Enter•Yes***. The display reads COMMAND COMPLETED and the preset is made.
- Press ***Load*** to return to the "Pile." Pressing the number that you selected for the preset will now instantly recall this keyboard configuration. Repeat this procedure for each performance preset you want to create, selecting a new number (from 1-8) for each.

Note: Performance presets are saved to disk only when you save the contents of memory as a bank. When you load a bank, any presets that were present at the time the bank was saved will be recreated.

CMD INST	DELETE INST EFFECT
	Press Command / Instrument / 6

This command deletes the Effect portion of an instrument. This is a handy "memory saver." If memory is tight and you're using several instruments through a single BANK effect, deleting the effect from each of the individual instruments in memory can free up enough memory to squeeze in that final track. Deleting the effect from an instrument trims about three blocks from its size.

- Select the ***Instrument•Sequence Track*** that has the effect that you want to delete.
- Select DELETE INST EFFECT.
- Press ***Enter•Yes***.
- The display shows COMMAND COMPLETED.

After performing this command, the instrument itself will contain no effect. However, the instrument can still use the BANK or ROM effects.

Effect Concepts

Section 5 — Effect Concepts

This section offers a basic overview of the concepts involved in selecting, creating, editing, and understanding Effects. For more detailed descriptions of the actual parameters, refer to the following section.

Understanding ASR–10 Effects

The ASR–10 has a powerful built in signal processor that can produce an extraordinary variety of effects. What is more important, its functions are integrated with, rather than added onto, the rest of the sampler. The flexible bus routing scheme and the extensive real time control give the ASR–10 its *dynamic* effects capability.

The ASR–10 offers continuous audio monitoring of the stereo audio inputs on two dedicated Audio Tracks. The audio inputs can be monitored through the Effects, and if desired, audio input can be sampled through the Effects. The ASR–10 also has the ability to resample sounds with effects in real time. For example: add reverb to a drum sample, resample it, then reuse the effects processor to further color your music.

The ASR–10's advanced digital signal processing system is based on the ENSONIQ Signal Processor (ESP) chip. The ESP is designed specifically for digital audio signal processing, and in the ASR–10 it works in tandem with a fourth generation version of the Digital Oscillator Chip (DOC IV, a.k.a. OTTO) and an external 16-bit digital-to-analog converter, to provide a very high-quality output signal.

The digital effects processing has been designed to complement the advanced performance features of the ASR–10, and many of the effects can have specific parameters modulated by various performance controls such as the modulation wheel, Poly-Key™ pressure, and others.

The effects are fully programmable and may be customized for particular applications. Effects are most often stored as part of an Instrument, although each bank has its own independent effect. Each of these types of effects is treated a little differently and will be described individually later in this section.

The ASR–10 has three distinct types of effects: Instrument Effects, Bank Effects, and ROM Effects, all of which behave in different ways.

Note: In this manual you may notice the term algorithm. An algorithm is the basic signal processing building block in the ASR–10. The word "effect" could be used instead of algorithm, but some algorithms can produce several sonic effects simultaneously. Each algorithm has a set of parameters used to control the effect(s) it produces.

INST (Instrument) Effects

Each instrument that you load into the ASR–10 contains an effect and a complete set of parameter values that determine how that effect will sound. The effect is present even if none of the voices in the instrument are routed through the effect (e.g. all voices are sent to the AUX 1, 2, or 3 destination busses), or FX= OFF, BANK, or ROM on the FX Select•FX Bypass page. Whenever you save an instrument, the effect settings are also saved with the instrument.

The current Instrument effect is displayed by pressing the ***FX Select•FX Bypass*** button and scrolling with the***Data Entry Slider*** or the ***Up/Down Arrow*** buttons until the display shows FX= INST, and edited by pressing ***Edit***, then ***Effects***.

Note: To delete an instrument's effect, use the Command/Instrument DELETE INST EFFECT Command.

Hint: When the FX Select•FX Bypass page is displayed, and FX= INST, pressing a loaded ***Instrument•Sequence Track*** button will display that instrument's effect. This feature provides a quick way to view which effect is assigned to each instrument.

BANK Effect

Each bank contains an effect and a complete set of effect parameters. When selected, this effect applies to all instrument/tracks (or their instrument voices) that are routed to BUS 1, 2, or 3. You can think of Instrument Effects as *individual* effects and Bank Effect as a *global* effect.

The Bank Effect is saved with each bank. It will remain unaffected until a new bank is loaded, unless it is specifically edited. Switching instruments within the bank will not select a new effect.

Whenever an independent effect file (one that is not attached to either an instrument or a bank) is loaded from disk into the ASR–10, it is placed in the Bank Effect location.

The Bank Effect is displayed by pressing the ***FX Select•FX Bypass*** button and scrolling with the Data Entry Controls until the display shows FX= BANK, and edited by pressing ***Edit***, then ***Effects***.

ROM Effect

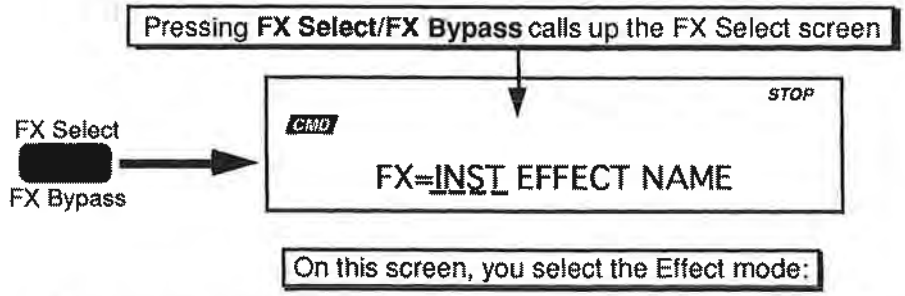
In addition to the Bank Effect, that can be changed by loading in a new effect from disk, or Command/Effect, COPY CURRENT EFFECT command (more on that later in the section), there are a large number of additional ROM Effects.

ROM Effects are similar to the Bank Effect in that they perform globally, affecting all instruments residing in memory. However, they differ in the fact that while the Bank Effect can be replaced, the ROM Effects are stored in *Read Only Memory* and will always reside in the ASR–10. Even after turning off the ASR–10, the ROM Effects will stay in memory.

The ROM Effects are displayed by pressing the ***FX Select•FX Bypass*** button and scrolling with the Data Entry controls. If you edit any of the parameters of a ROM effect (on the Edit/Effects page), that effect is instantly copied into the Bank Effect location.

Hint: To return to the ROM Effect location after editing any parameters (after the ROM effect has become the Bank Effect), from the FX Select•FX Bypass page, press ***Cancel•No***. This function can also serve as a handy "MIDI panic button" (all notes off) to stop any droning notes received via MIDI.

Selecting Effects



FX=	OFF	INST	BANK/ROM#
	When OFF is selected, all sounds are dry, regardless of their output setting.	1 2 3 4 5 6 7 8 When INST is selected, the instruments control the effect. In this mode, each time an instrument is selected, its related effect is loaded into the ASR–10. Any stacked instruments will use the primary instrument's effect.	Both BANK and the ROM effects are GLOBAL effects. When selected, they become the effect for all instruments. Each instrument still has control over its Output Bus and Mix controls.

The main parameter of the FX Select/FX Bypass page is the effects selector. Changing this parameter causes a new effect to be selected, which in turn changes the type of parameters that will be available on the Edit/Effects page. This parameter may also change the Effect Type, which will affect how individual instruments within the internal memory will interact with the effect (see the preceding discussion on effect types). Selecting a new effect will automatically set all of the effect parameters to their default values for the new effect. The ROM Effects, those that always reside within the internal memory of the ASR–10, are selected here.

ROM Effects List

These are the available ROM Effects within the ASR–10:

HALL REVERB	Hall, left to right (transverse) reflections
44KHZ REVERB	A hall reverb utilizing the 44KHZ playback rate
ROOM REVERB	A reverb featuring a shorter, room-like decay
DUAL DELAYS	A dual delay with separate left and right delay times
44KHZ DELAYS	A dual delay utilizing the 44KHZ playback rate
CHORUS+REVRB	Four-voice chorus combined with reverb
PHASER+REVRB	Phase shifter combined with reverb
FLANGER+REV	Flanger combined with reverb

ROT. SPKR+REV	Rotating speaker simulator with reverb and distortion
CHOR+REV+DDL	Chorus with reverb plus digital delay
CMP+DIST+REV	Compression, Distortion, and Reverb. Great for guitar and amp simulation with acoustic feedback
DIST+CHO+REV	Distortion with chorus and reverb
WAH+DIST+REV	"Volume-Wah" effect with distortion and reverb
SMALL ROOM	An ambient room reverb
LARGE ROOM	A room reverb with larger ambience
HALL REVERB2	This hall reverb offers a large acoustic space
SMALL PLATE	A tight sounding plate reverb
LARGE PLATE	Simulates a larger plate reverb
REVRSE REVRB	A reverb that gradually increases, simulating a backwards sound
REVRSE REVB2	Like REVRSE REVRB, this algorithm will retrigger by a controllable input signal level
GATED REVERB	The popular gated reverb sound
NLIN REVRB 1	Non linear reverb with a short duration time
NLIN REVRB 2	Non linear reverb with a long duration time.
NLIN REVRB 3	Similar to NLIN REVRB 1, with less stereo movement
MULTITAP DDL	Four independent controllable delays
EQ+DELAY LFO	A stereo delay with LFO modulation
VCF+DISTORT	Combines a voltage control filter and a raspy distortion.
GUITAR AMP1	The warm sound of a Hard Rock guitar amp
GUITAR AMP2	The "bluesy" sound of a guitar amp
GUITAR AMP3	An inverse expander with a bright distortion amp for lead sounds
SPKR CABINET	Simulates an open back speaker
TUNABLE SPKR	An EQ controllable speaker sound
EQ+CHOR+DDL	EQ with chorus plus digital delay
EQ+VIBR+DDL	EQ with vibrato plus digital delay
EQ+FLNGR+DDL	EQ with flanger and digital delay
EQ+TREM+DDL	EQ with tremolo and digital delay
PHASER+DDL	Phaser with a digital delay
8-VOICE CHOR	A rich chorus with eight different voices
PITCH SHIFT	Splicer type pitch shifter
PITCH+DDL	A pitch shifter and a digital delay
FAST PITCHSH	Pitch shifter designed for pitch correction
EQ+CMPRESSOR	EQ with a compressor
EXPANDER	Performs downward expansion of input signals
KEYED EXPNDR	An expander that is triggered from another source
INVRS EXPNDR	Creates sustain by expanding low signal levels
DE-ESSER	Compresses sibilant frequencies
DUCKER	One signal controls the volume of another signal by compression
RUMBLE FILTR	High pass filter in cascade with a low pass filter
PARAM EQ	Minimum phase four band parametric EQ
VAN DER POL	Adds synthetic high harmonics, brightening the overall sound

Sample Rate

Each Effect has a defined number of voices that it can play, based on its sample rate. To view the current sample rate, from the FX Select screen, press the ***Left Arrow*** button. The display shows:

Pressing the ***Right Arrow*** button will return you to the FX Select screen.

Note: Changing the Effect could change the sample rate that will be viewed in sample mode. A lower sample rate gives you a longer sample time, because it takes fewer samples per second. A higher sample rate allows you to sample with increased fidelity (because it takes more samples per second). Changing the sample rate also affects the polyphony:

If the sample rate is...	then polyphony is...	and the sample time is...
30 kHz	31 voices	longer
44 kHz	23 voices	shorter

Note: When the optional DI-10 Digital Interface is installed, the Digital Output will provide digital audio output of the Main Output mix, if the current effect uses a 44.1 KHz sample rate

Instruments and Banks

The complete effects setup, including the values of all effect parameters, is saved when you save an instrument. It is *also* saved with each bank. The ASR–10 is smart about switching effects, since all sound must stop for an instant when it changes effects.

When are New Effects Loaded Into the ESP chip?

- When you select an instrument from one of the ***Instrument•Sequence Track*** buttons with FX=INST on the FX Select/FX Bypass page, the effect saved in that instrument will be loaded into the ESP, and you will hear the sound with its associated effect.

- When you load a new Bank, a new Bank Effect is loaded into the ESP.

- When you load an individual effect (one that is not part of a Bank or an Instrument), it is loaded into the ESP and becomes the new Bank Effect.

- When you select or layer instrument/tracks with FX=BANK or FX=ROM on the FX Select/FX Bypass page, the effect is *not* changed.

Whenever a new effect is loaded into the ESP, the audio output will pause briefly, allowing the instructions that create the new effect algorithm to be loaded into the ESP. If an effect differs only by variation in parameter values, then this pause will not occur.

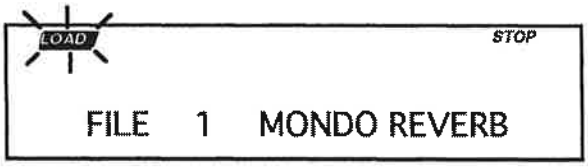
Loading and Saving ASR–10 Effects

Loading an Effect File

You can load an effect from disk into the internal memory of the ASR–10. The effect will load into the Bank effect location, replacing whatever effect was there previously.

First, insert a disk containing one or more effect files into the disk drive.

- Press ***Load***. The LOAD indicator flashes.
- Press ***Effects***. The display looks something like this:



When the LOAD indicator is flashing, the ASR–10 is showing you disk files. Pressing the ***Up/Down Arrow*** buttons takes you through the effect files on disk. If there are none, the display will read NO EFFECT FILES.

Whenever a disk file is displayed as above, you can press the ***Left*** or ***Right Arrow*** button to see the size of that file in *Blocks* (a Block is 256 samples; 4 Blocks=1K samples). Press the ***Left*** or ***Right Arrow*** button again to return to the file name.

- Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to view the various effect files on the disk. Each file has its own File Number.
- Find the effect you want to load, and press ***Enter•Yes***. The ASR–10 will load that effect into the BANK effect slot. When finished, the display will read DISK COMMAND COMPLETED.

Saving an Effect to Disk

Use the SAVE BANK EFFECT command to save an effect to disk. If you wish to save, as a separate effects file, either a ROM effect or an INST effect, you must first copy the effect into the BANK effect using the COPY CURRENT EFFECT parameter on the Command/Effects page (see *Section 6 — Effect Parameters*) and then save it.

- Insert a formatted disk into the drive.
- Press ***Command***, then press ***Effect.***
- Press the ***Left*** or ***Right Arrow*** button until the display reads SAVE EFFECT.
- Press ***Enter•Yes.***
- Edit the effect name (if necessary):

The display shows the current name of the effect, with a cursor (underline) beneath the first character. If you want to give the effect a new name, do so at this time. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the underlined character, then press the ***Left/Right Arrow*** buttons to move the underline. Repeat until the display shows the name you want. If you are saving a ROM effect, the name will not be able to be edited (if you don't need to rename the effect, just skip this step).

- Press ***Enter•Yes.*** The display will read SAVING <FILE NAME> while the effect is being saved.
- If there is already an effect file with the same name on the disk, the display will ask DELETE OLD VERSION? Press ***Enter•Yes*** to save the effect, replacing the one on the disk. Or press ***Cancel•No*** to abort the procedure.
- If there is not enough free space on the disk, the display will show NOT ENOUGH DISK SPACE. Save the effect to another disk (or delete some files from the disk).

Programming Effects

The ASR–10 effects are highly programmable. Each effect has a wide variety of parameters for editing. The FX Select/FX Bypass page contains the parameter for selecting which effect will be used. This page controls how all of the other effect screens (found on the Edit/Effects page) will be configured and displayed. When this parameter is changed, a new effect is selected which causes several important things to occur:

- a new effect is loaded, causing a brief pause in the audio output
- the number of voices and the system sample rate may change
- the effect parameters are redefined for the particular effect selected
- the effect parameter values are reset to their default settings

Effect Variations

The first parameter screen is similar for all of the effects (the display shows VAR=). It selects one of four Effect Variations, which feature various "preset" patches, letting you quickly access several set-ups within the same effect. Each of these four variations is individually editable and can act as four effect edit buffers. Each Effect Variation can have two different Modulation Sources, Destination parameters, and defined modulation ranges (for a total of eight different modulatable parameters).

The Effects Busses

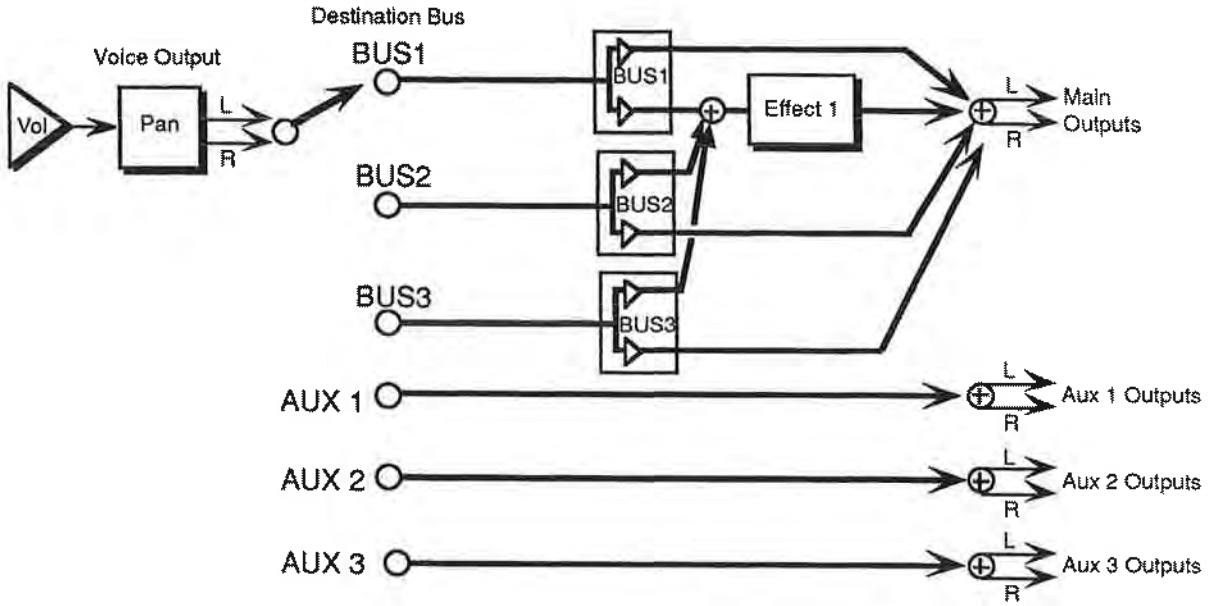
The output of every voice in the ASR–10 is assigned to a stereo bus. A bus, like the bus of a mixing board, mixes together all the voices assigned to that bus into a single stereo pair. Of the six busses on the ASR–10, three are inputs into the signal processor (BUS1, BUS2, and BUS3), and three are direct paths to the AUX outputs on the optional Output Expander, bypassing all effect processing (dry). The Destination Bus assignment for each WaveSample is set on the Edit/Amp page. The voice settings in the sound can be overridden for each bank from the FX Select/FX Bypass page.

Effects Mixing

All effects have separate mixing controls for the BUS1, BUS2, and BUS3 (when not used as a DRY path) busses. They are found on three screens within the Edit/Effects page, and will have slightly different wording depending on the effect they're contained within.

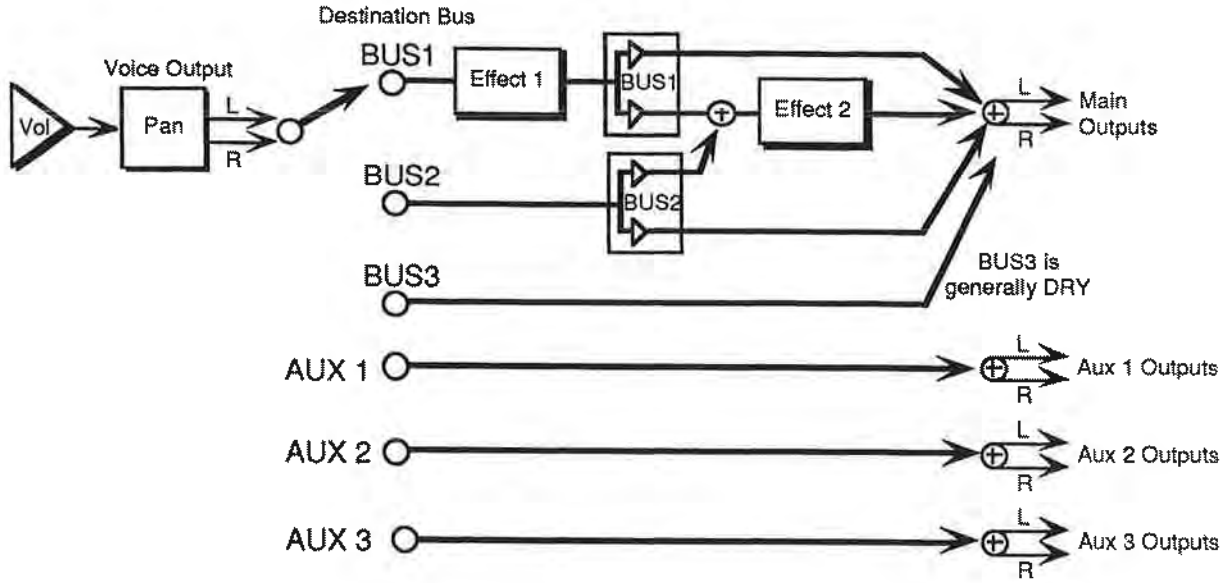
When an effect having a single processing function (such as reverb only) is selected, both busses BUS1 and BUS2 are routed to it. When using a dual or multi-effect, BUS1 will generally route the signal through both or all effects, with BUS2 routing only through the second effect. In most cases, the BUS3 bus is reserved as a DRY path to the main outputs.

Single Function Effect Mixer



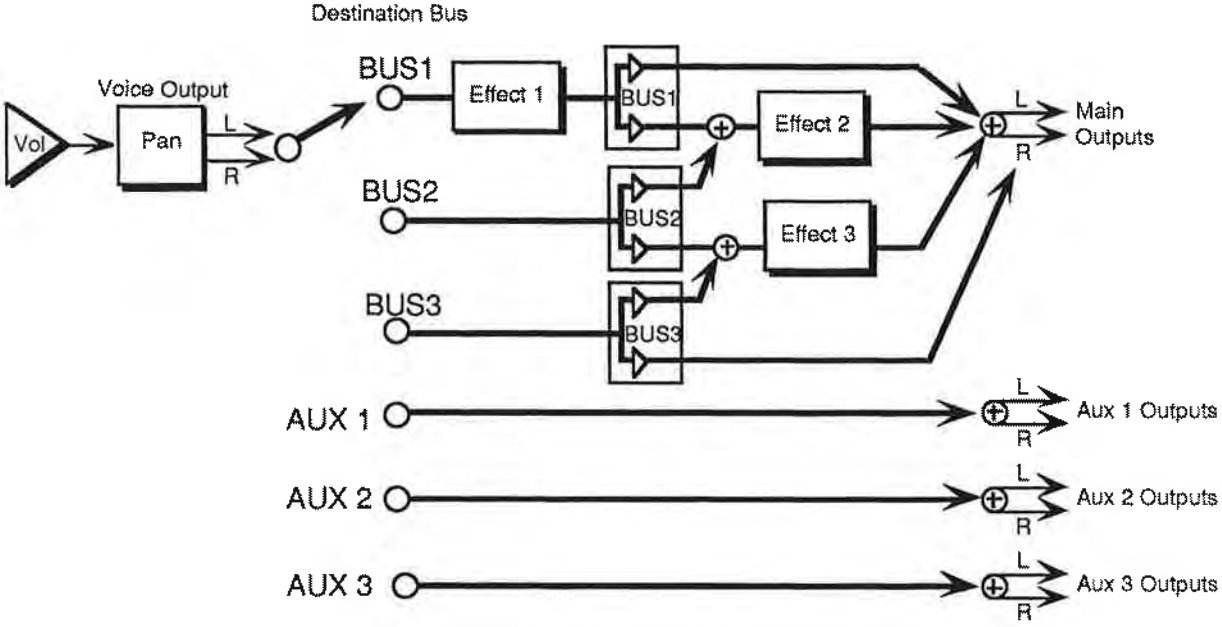
The above diagram shows the effects busses and the output mixing. Every voice is assigned to one of the four stereo busses, which go around or through the effects processing. The heavy lines are stereo paths. Some effects, such as HALL REVERB, also have a Dry/Wet mix of BUS3 into Effect 1.

Dual Function Effect Mixer



When the selected effect is a *combined effect* that has more than one signal processing function (such as chorus and reverb), the BUS 1 bus feeds Effect 1 and Effect 2, and the BUS2 bus feeds Effect 2 only. Within effects there may also be Dry/Wet mixes (such as the Chorus Mix within the CHORUS+REVERB effect).

Triple Function Effect Mixer



Some effects, such as DIST+CHO+REV, use the above mixing scheme. Effect 1 and Effect 2 usually have a Dry/Wet mix within them. This mixer can be configured as three independent effects, or three effects in a series.

Performance Control of Tracks When Using Bank Effects

When using a Bank Effect, all the instruments in the internal memory are always routed through the same global effect. In order to gain some individual control for each instrument, two parameters on the Edit/Track page determine how each instrument will interact with the Bank Effect.

- OUT — This parameter determines which of the Busses the instrument will be sent through. The options are: WAVESAMPLE, BUS1, BUS2, BUS3, AUX1, AUX2, and AUX3. By using this parameter you can give individual instruments different Dry/Wet or multiple effect mixes. The WAVESAMPLE output bus is not available for Audio Tracks.

- EFFECT MOD CONTROL — This parameter determines whether the controllers of a particular instrument will be routed to control the effect. This is an important setting when using a Bank Effect: since all instruments are routed through the same effect, it could be confusing to have multiple instruments sending conflicting controller information to the effect.

Predefined Effect Modulation Parameters

Several of the effects allow real time control of a single predefined parameter within the effect, such as the Reverb Mix or Flanger Feedback. Each of these effects will have its own Modulator Select screen, where the modulator and mod amount are selected. Each Effect Variation can have a different assigned Modulation Source and Amount (for a total of four different modulatable parameters). This screen will look different from effect to effect, but it will always be the last screen on the Edit/Effect page. The screen has two parameters:

- Mod Source — This selects the source for modulating the defined parameter within the selected effect.
- Mod Amount (Mix) — This sets the amount the Mod Destination will be modulated by the Mod Source.

The following modulation sources can alter the effects in performance:

Mod Source	Modulation effect derived from
WHEEL	the value of the mod wheel
PBEND	the value of the pitch wheel (50% output at rest)
PRESS	the channel pressure value for the selected track
PEDAL	the value of the Pedal/CV input
XCTRL	uses the value of the assignable external MIDI controller, set on the Edit/System•MIDI page
KBD	the number of the last key played
VEL	average velocity of all keys played simultaneously
KEYDN	ON when any keys are held down, otherwise OFF
SUSTN	ON when sustain pedal is held down; otherwise OFF
FSW-L	uses the left foot switch of the optional SW-10 Foot Switch; ON when held down, otherwise OFF (Edit/System•MIDI LEFT FOOT SW must be set to FX MODSRC for this to function as an FX modulator)
PATCH	four discrete values, ranging from off (no Patch Select Foot Switches held) to full ON (both Patch Select Foot Switches held)
OFF	no modulation

Selectable Effect Modulation Parameters

Most of the effects allow real time control of selectable parameters and share common modulation control parameters. Within these effects, each Effect Variation can have two different Modulation Sources, Destination parameters, and defined modulation ranges (for a total of eight different modulatable parameters). The Modulation Control Parameters are always the last eight parameters on the Edit/Effect page:

- **MOD SRC 1 — Modulation Source 1**
- **MOD SRC 2 — Modulation Source 2**

Sources: WHEEL, PBEND, PRESS, PEDAL, XCTRL, KBD, VEL, KEYDN, SUSTN, FSW-L, PATCH, or OFF

These parameters select the modulation sources used to modulate the parameter Destinations. Each Effect Variation has a choice of two discrete mod sources.

- **DST 1 — Mod1 Destination Parameter**
- **DST 2 — Mod2 Destination Parameter**

Ranges: various (selected parameter name)

The Destination parameters select which effect parameters will be modulated by the modulation sources. The choice of parameters varies depending on the selected effect. Any parameter within an algorithm can be selected. Each Effect Variation has a choice of two different mod destinations.

Hint: Once the destination has been defined, you can toggle between the DST (Destination) parameters and the actual parameter that is selected to be modulated, by alternate presses of the ***Enter•Yes*** button. When the parameter being modulated is displayed, you can watch the actual value change as the mod source output changes. You cannot edit a parameter when it is assigned as a modulator destination. Disabling modulation of a parameter will restore its saved value.

- **MIN 1 — Mod 1 Param Range Minimum**
- **MAX1 — Mod 1 Param Range Maximum**
- **MIN 2 — Mod 2 Param Range Minimum**
- **MAX2 — Mod 2 Param Range Maximum**

Ranges: 0 to 99

These four parameters set the minimum and maximum amount (based on a percentage of the selected parameter's range) that the Mod Destination will be modulated by the Mod Source. Inverting the amounts will also invert the Mod effect.

Remember

In order for the effect modulation to work, the EFFECT MOD CONTROL parameter on the Edit/Track page must be set to ON for the selected ***Instrument•Sequence Track***.

Effect Parameters

Section 6 — Effect Parameters

This section offers detailed descriptions of the commands and parameters used in selecting, creating, and editing effects. For a basic overview of the concepts involved, refer to the previous section.

Command/Effects Page

The commands on this page allow you to copy and save effects.

CMD EFFECTS	SAVE BANK EFFECT
	Press Command / Effects / 0

This parameter is used to save the BANK effect to disk (or SCSI device) as a separate file that can be loaded at any time. Only the BANK effect can be saved directly to disk. Conversely, when you load an effect file from disk, it is always loaded into the BANK effect location. If you wish to save either a ROM effect or an INST (instrument) effect as a separate effects file, you must first copy the effect into the BANK effect (see below) and then save it.

- Select SAVE BANK EFFECT.
- Press ***Enter•Yes***. The display shows the file name associated with the current BANK effect. You can rename the effect using the ***Arrow*** buttons and the ***Data Entry Slider*** if you wish.
- Press ***Enter•Yes*** to save the effect file to disk.

CMD EFFECTS	COPY CURRENT EFFECT
	Press Command / Effects / 1

This command is used to copy an effect either from the bank to an instrument, or the other way around. It can also be used to copy one of the ROM effects into an instrument or to the BANK effect. The effect that is copied (the source effect) is always the one currently selected on the FX Select•FX Bypass page.

- Select COPY CURRENT EFFECT.
- Press ***Enter•Yes***. The display shows the DIRECTION parameter. Using the ***Up/Down Arrow*** buttons, select either COPY TO BANK to take the effect from the currently selected instrument and copy it into the BANK effect, or COPY TO INST to take the effect currently displayed on the FX Select•FX Bypass page and copy it to one of the instruments in the internal memory.
- Press ***Enter•Yes***. If you've selected COPY TO BANK in the previous step, the display will briefly read COMMAND COMPLETED and the procedure will be done. However, if COPY TO INST was selected, use the ***Instrument•Sequence Track*** buttons to select which instrument you wish to copy the BANK effect into. Once the destination instrument has been selected, press ***Enter•Yes***.

Edit/Effects Page

Each of the effects has multiple screens containing a particular set of parameters associated with that effect. Press the ***Edit*** button, followed by the ***Effects*** button to access these parameters.

Effect Variations

The first parameter screen is similar for all of the effects (the display shows VAR=). It selects one of four Effect Variations, which feature various "preset" patches, letting you quickly access several set-ups within the same effect. Each of these four variations is individually editable and can act as four effect edit buffers. Each Effect Variation can have two different Modulation Sources, Destination parameters, and defined modulation ranges (for a total of eight different modulatable parameters). Subsequent screens are variable, and contain pertinent parameters that provide the flexibility for customizing the effects.

ROM-01 HALL REVERB
ROM-02 44KHZ REVERB
ROM-03 ROOM REVERB

These effects algorithms provide studio-quality stereo reverb. The BUS1, BUS2, and BUS3 parameters can be routed into the effect with different Dry/Wet mixes. HALL REVERB is a reverb with a long decay, featuring left to right (transverse) reflections. 44KHZ REVERB is a hall reverb utilizing the 44KHZ system sample rate (polyphony is reduced to 23 voices). ROOM REVERB features a shorter, room-like decay.

The parameters available in these three effect algorithms are:

DECAY TIME
Range: 0 to 99
Controls the amount of time it takes for the reverberation to decay to a low level (-60 dB) after the input signal stops.

LOW FREQ DECAY — Low Frequency Decay
Range: -99 to +99
This control will boost or cut the rate at which low frequencies will decay.

DIFFUSION
Range: 0 to 99
This parameter determines whether the early reflections will appear as a series of discrete echoes (lower values) or will be more diffused (higher values).

HI FREQ DAMPING — High Frequency Damping
Range: 0 to 99
This parameter controls the rate of attenuation of high frequencies in the decay of the reverberation. As natural reverb decays, some high frequencies tend to get absorbed by the environment. Increasing the value of this parameter will filter out increasing amounts of high frequency energy.

HI FREQ BANDWIDTH — High Frequency Bandwidth
Range: 0 to 99
The High Frequency Bandwidth parameter acts as a low pass filter on the signal going into the reverb, controlling the amount of high frequencies that will pass into the effect. The higher the setting, the more high frequencies are allowed to pass.

DETUNE RATE
Range: 0 to 99
This parameter controls the LFO rate of detuning incorporated within the reverb. Detuning creates a slight pitch shift into the reverberated signal, giving it a more natural sounding decay by breaking up resonant nodes.

DETUNE AMT — Detune Amount
Range: 0 to 99
This parameter controls the depth of the detuning, that is, how much the pitch will change. Low values yield a metallic sound. Some voices sound best with very low values.

PRE-DELAY TIME
ROOM and HALL REVERB Range: 0 to 500
44KHZ REVERB Range: 0 to 340
This parameter creates a delay in the amount of time it takes before the original signal is fed into the reverb. The range is based in milliseconds.

ROM-04 DUAL DELAYS
ROM-05 44KHZ DELAYS

A professional quality stereo digital delay. DUAL DELAYS uses the standard 30KHZ (31 voice) system sample rate, while 44KHZ DELAYS utilizes the 44KHZ system sample rate (polyphony is reduced to 23 voices). Both algorithms have identical parameters. Both BUS1 and BUS2 can have different Mix, Feedback, and Delay Time settings. BUS3 is dry.

BUS1 (and 2) DELAY MIX
Range: 0 to 99
This parameter controls the mix between the original signal and the delay. Setting this parameter to 0 will allow only the unprocessed signal to be heard, while a setting of 99 will eliminate it completely, with only the delay portion remaining.

BUS1 (and 2) FEEDBACK
Range: 0 to 99
Determines the amount of signal that will be fed from the output of the right delay back into the input, increasing the number of repeats in the delay.

BUS1 (and 2) DELAY TIME — Left and Right
Dual Delays Bus1 Range: 0 to 2000
44 KHZ Delays Bus1 Range: 0 to 1300
Bus2 Range: 0 to 150
Determines the amount of the amount of time between delays. Each value increases the delay time by 1 millisecond. The Left and Right channels are independently adjustable.

ROM-06 CHORUS+REVRB

This effect combines a four-voice chorus with the standard reverb. Assign a voice to BUS1 to get both chorus and reverb, or use BUS2 for reverb only. Use BUS3 to bypass the effect (dry).

BUS1 REV AFTER CHOR — Reverb After Chorus
Range: 0 to 99
This parameter determines how much of the signal that is sent to BUS 1 (the chorus) will also go to the reverb — it is a Reverb Mix for the chorused signal.

BUS2 REVERB MIX
Range: 0 to 99
This parameter controls the Dry/Wet mix of the Reverb for voices sent to BUS2.

REVERB DECAY TIME
LOW FREQ DECAY — Low Frequency Decay
DIFFUSION
HI FREQ DAMPING — High Frequency Damping
HI FREQ BANDWIDTH — High Frequency Bandwidth
REVERB DETUNE RATE
REVERB DETUNE AMT
See the description under HALL REVERB.

EARLY RFLECTION LEV — Early Reflection Level
Range: 0 to 99
Controls the level of pre-echoes of the input signal added directly to the reverb output.

CHORUS RATE
Range: 0 to 99
This parameter controls the four rates of modulation of the delay time of the chorus. The delay modulation produces vibrato and tremolo.

CHORUS AMT — Chorus Amount
Range: 0 to 99
This parameter controls the depth of vibrato and tremolo.

CHORUS MIX
Range: 0 to 99
This parameter controls the Dry/Wet mix within the chorus itself.

CHORUS FEEDBACK
Range: -99 to +99
This parameter controls the amount of feedback applied to the chorus. The sign of the value determines the polarity of the feedback. Feedback accentuates the complexity of the "swept" signal.

ROM-07 PHASER+REVRB

A 12-pole phase shifter with reverb. Assign a sound to BUS1 for phaser plus reverb; to BUS2 for just reverb; and to BUS3 to bypass the effect (dry).

BUS1 REVERB MIX
Range: 0 to 99
This parameter controls the Dry/Wet mix of the Phaser into the Reverb.

BUS2 REVERB MIX
Range: 0 to 99
This parameter controls the Dry/Wet mix of the Reverb.

REVERB DECAY TIME
LOW FREQ DECAY — Low Frequency Decay
DIFFUSION
HI FREQ DAMPING — High Frequency Damping
HI FREQ BANDWIDTH — High Frequency Bandwidth
REVERB DETUNE RATE
REVERB DETUNE AMT
See the description under HALL REVERB.

EARLY RFLECTION LEV — Early Reflection Level
See the description under Chorus+Reverb.

PHASE CENTER
Range: 0 to 99
This parameter controls the center of the phaser width.

PHASE RATE
Range: 0 to 99
This parameter controls the rate of modulation of the phaser poles.

PHASE WIDTH
Range: 0 to 99
This parameter controls the amount of modulation applied to phaser poles.

PHASE FEEDBACK
Range: -99 to +99
This parameter controls the amount of feedback applied to the phaser. The sign of the value determines the polarity of the feedback.

PHASE DEPTH
Range: 0 to 99
This parameter controls the depth of the notches created by the phasing.

ROM-08 FLANGER+REV

The FLANGER+REV algorithm is similar to the Chorus+Reverb algorithm with a single LFO. Assign a voice to BUS1 to get the flanger plus reverb, and use BUS2 for reverb only. BUS3 is dry.

BUS1 REVERB MIX
Range: 0 to 99
This parameter controls the Dry/Wet mix of the Flanger into the Reverb.

BUS2 REVERB MIX
Range: 0 to 99
This parameter controls the Dry/Wet mix of the Reverb for signals sent to BUS2.

REVERB DECAY TIME
LOW FREQ DECAY — Low Frequency Decay
DIFFUSION
HI FREQ DAMPING — High Frequency Damping
HI FREQ BANDWIDTH — High Frequency Bandwidth
REVERB DETUNE RATE
REVERB DETUNE AMT
See the description under HALL REVERB.

EARLY RFLECTION LEV — Early Reflection Level
See the description under Chorus+Reverb.

FLANGE CENTER
Range: 0 to 99
This parameter controls the center of the flanger width.

FLANGE RATE
Range: 0 to 99
This parameter controls the rate of modulation of the flanger effect.

FLANGE WIDTH
Range: 0 to 99
This parameter controls the range of the high to low frequency sweep in the flanger effect.

FLANGE FEEDBACK
Range: -99 to +99
This parameter controls the amount of feedback applied to the flanger. The sign of the value determines the polarity of the feedback.

FLANGE DEPTH
Range: 0 to 99
This parameter controls the depth of the notches created by the flanging effect.

ROM-09 ROT. SPKR+REV

A rotating speaker simulation with reverb and distortion. Assign a voice to BUS1 to get both rotary speaker with distortion and reverb, and use BUS2 for reverb only. BUS3 is a dry bus (no effect). There are selections on the Variations screen for the Rotary Speaker without Distortion.

BUS1 REV AFTER ROTO — Reverb After Rotary Speaker
Range: 0 to 99
This parameter controls the Dry/Wet mix of the Rotary Speaker.

BUS2 REVERB MIX
Range: 0 to 99
This parameter controls the Dry/Wet mix of the Reverb from signals sent to BUS2.

REVERB DECAY TIME
HI FREQ DAMPING — High Frequency Damping
DIFFUSION
See the descriptions under Hall Reverb.

ROTOR FAST
Range: 0 to 99
Determines the rate of the rotary speaker when in the "Fast" setting. The higher the value, the faster the rate.

ROTOR SLOW
Range: 0 to 99
Determines the rate of the rotary speaker when in the "Slow" setting. ROTOR SLOW determines the manual level for the rotary speaker rate when FAST-SLOW SWITCH= OFF, or when the selected modulator is at zero output level. Again, the higher the value, the faster the rate.

ROTOR INERTIA
Range: 0 to 99
Determines how long it will take for the rotor effect to speed up and slow down after switching from fast to slow or vice versa. Adjust this parameter to simulate the effect of the rotary speaker gradually picking up speed.

FAST-SLOW SWITCH
Determines how the rotary speaker will switch between slow and fast speeds. The behavior of the switch accurately reflects an actual rotary speaker, taking time to speed up or slow down, based on the value of the ROTOR INERTIA parameter above. Any effect modulator can act as the FAST-SLOW SWITCH. How the modulators will switch the rotor speed fall into two categories:

- WHEEL, PBEND, PRESS(ure), PEDAL, XCTRL, KBD, and VEL — These modulation sources act like a switch to turn on or off the fast rotor speed. To reverse the polarity of the switch, set ROTOR SLOW faster than ROTOR FAST.
- KEYDN, SUSTN, FSW-L, PATCH — These modulation sources toggle the rotor speed between slow and fast. Every time the modulation source moves from zero in a positive direction, the rotating speaker effect changes speeds from slow to fast or fast to slow.

DIST GAIN IN/OUT — Distortion Input and Output Gain
In and Out Range: 0 to 99
The IN parameter determines the input signal gain into the amplifier simulation, creating a tube-like overdrive.

The OUT parameter controls the *output* of the amplifier distortion. There is a (fixed) clean path in parallel with the distortion. Therefore, to eliminate distortion, set this parameter to zero.

SPEAKER GAIN
Range: 0 to 99
Controls the level of the signal (distorted or not) applied to the rotary speaker. High gain will simulate a broken speaker. Also use higher settings when using little or no distortion, because distortion raises the signal level to the speaker.

DISTORTION FILTER
Range: 0 to 99
A three-pole, low pass filter following the amplifier simulation. This parameter controls the filter cutoff resulting in a brighter sound (for higher values) or a darker sound (low values).

ROM-10 CHOR+REV+DDL

Chorus with digital delay and reverb. Assign a voice to BUS1 for Chorus and Reverb, BUS2 for Reverb only, and BUS3 for Digital Delay (DDL).

BUS1 REV AFTER CHOR — Reverb After Chorus
BUS2 REVERB MIX
REVERB DECAY TIME
LOW FREQ DECAY — Low Frequency Decay
HI FREQ DAMPING — High Frequency Damping
REVERB DETUNE RATE/AMT
CHORUS RATE/AMT
CHORUS MIX
CHORUS FEEDBACK
See the descriptions under Chorus+Reverb.

BUS3 DDL MIX
Range: 0 to 99
The BUS3 DDL Mix determines the Dry/Wet mix of the Digital Delay.

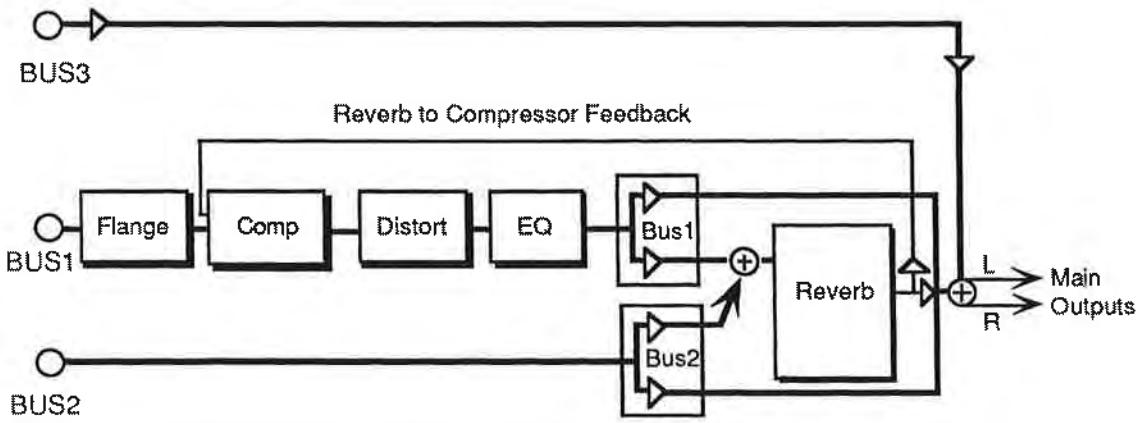
DDL TIME L/R
Range: 0 to 1500
This parameter determines the delay time for the left and right channels independently. It is adjustable from 0 to 1500 in 1 millisecond increments.

DDL FEEDBACK
Range: 0 to 99
The DDL Feedback parameter controls the amount of feedback applied to the Digital Delay.

ROM-11 CMP+DIST+REV

A screaming guitar-and-amp simulator that features not only compression, distortion, and reverb, but flanging and EQ as well. BUS1 routes the signal through each of these effect processors, while BUS2 is used for reverb only. BUS3 is dry.

CMP+DIST+REV Signal Routing



BUS1 REV AFTER DIST — Reverb After Distortion
Range: 0 to 99
This parameter controls the Dry/Wet mix of the Distortion.

BUS2 REVERB MIX
REVERB DECAY TIME
HI FREQ DAMPING — High Frequency Damping
See the descriptions under HALL REVERB.

COMPRESSOR THRESHLD
Range: 0 to 99
This parameter controls the threshold level for the compressor. As the input signal dies away below the threshold, the compressor will increase the gain of the system, causing feedback to increase as well. Normal compression is "72."

DIST GAIN IN and OUT
Ranges: 0 to 99
These two parameters control the levels going into and coming out of the distortion effect. For more distortion, use a high input level and turn the output level down to keep the volume under control. For less distortion, use a low input level and a higher output level.

DIST LPF/HPF — Distortion Low Pass Filter/High Pass Filter
Range: 0 to 99
The DIST LPF parameter filters out high frequencies after the distortion signal path. The lower the value, the less high frequencies pass through. The DIST HPF parameter filters out low frequencies after the distortion signal path. The higher the value, the less low frequencies pass through.

DIST FLANGE RATE
Range: 0 to 99
This parameter controls the rate of modulation of the notches due to the flanger effect. A setting of 0 effectively turns the flanger off.

SYSTEM FEEDBACK
Range: -99 to +99
This parameter controls the amount of signal applied from the output of the reverb back into the input of the compressor. The sign of the value determines the polarity of the feedback.

Note: Since the feedback level is taken from BUS1, if BUS1 is set to +0 or a low value, feedback will have little or no consequence. Also, it is important to note that since the signal is being taken from the reverb, that is also fed by BUS2, voices sent to BUS2 will enter the feedback loop along with the BUS1 signals.

ROM-12 DIST+CHO+REV

A brighter guitar-and-amp simulator that features distortion, chorus, and reverb, but without the compressor. The three effects can be run in a serial or parallel configuration, depending on how the three BUS parameters are set. This effect uses the triple function effect mixing scheme.

BUS1 DIST INTO CHOR MIX — Distortion Into Chorus Mix
Range: 0 to 99
This parameter determines how much of the output from the distortion bus will be fed into Bus2 for added chorusing. Setting this parameter to 0 will cause the output from the distortion bus to go directly to the main outputs.

CHOR INTO REV MIX — Chorus Into Reverb Mix
Range: 0 to 99
This parameter determines how much of the output from the chorus bus will be fed into Bus3 for added reverb. Setting this parameter to 0 will cause the output from the chorus bus to go directly to the main outputs.

BUS3 REVERB MIX
Range: 0 to 99
Sets the Dry/Wet mix of the reverb.

DIST GAIN IN and OUT
Ranges: 0 to 99
See the description under CMP+DIST+REV.

DIST FILTER FC — Distortion Filter Cutoff Frequency
Range: 0 to 99
This determines the filter cutoff frequency (brightness) of the distortion.

DIST FILTER Q — Distortion Resonant Peak
Range: 1 to 25
This parameter determines the level of the resonant peak at the filter cutoff point. While the FILTER FC parameter determines where (at what frequency) this peak will occur, the Q setting controls the presence of the peak.

DISTORTION MIX
Range: 0 to 99
Determines the Dry/Wet, or in this case, dirty/clean mix of the signal. A value of 0 yields a clean signal; 99 yields an all distortion signal.

REVERB DECAY TIME
LOW FREQ DECAY — Low Frequency Decay
HI FREQ DAMPING — High Frequency Damping
HI FREQ BANDWIDTH — High Frequency Bandwidth
REVERB DETUNE RATE/AMT
See the descriptions under HALL REVERB.

EARLY RFLECTION LEV
CHORUS RATE/AMT
CHORUS MIX
See the descriptions under CHORUS+REVERB.

ROM-13 WAH+DIST+REV

A guitar and amp simulator, featuring distortion, compression, reverb, and a resonant filter following the amplitude of the signal. Assign a sound to BUS1 for Wah distortion and reverb; use BUS2 for reverb only. BUS3 is dry.

BUS1 REV AFTER DIST — Reverb After Distortion
Range: 0 to 99
This parameter controls the Dry/Wet mix of the Distortion.

BUS2 REVERB MIX
REVERB DECAY TIME
REVERB DAMPING — Reverb Damping
See the descriptions under HALL REVERB.

COMPRESSOR THRESHLD — Compressor Threshold
DIST GAIN IN and OUT
See the descriptions under CMP+DIST+REV.

WAH FILTER CENTER
Range: 0 to 99
Determines where the resonant peak of the wah will occur. Higher values create a higher pitched wah.

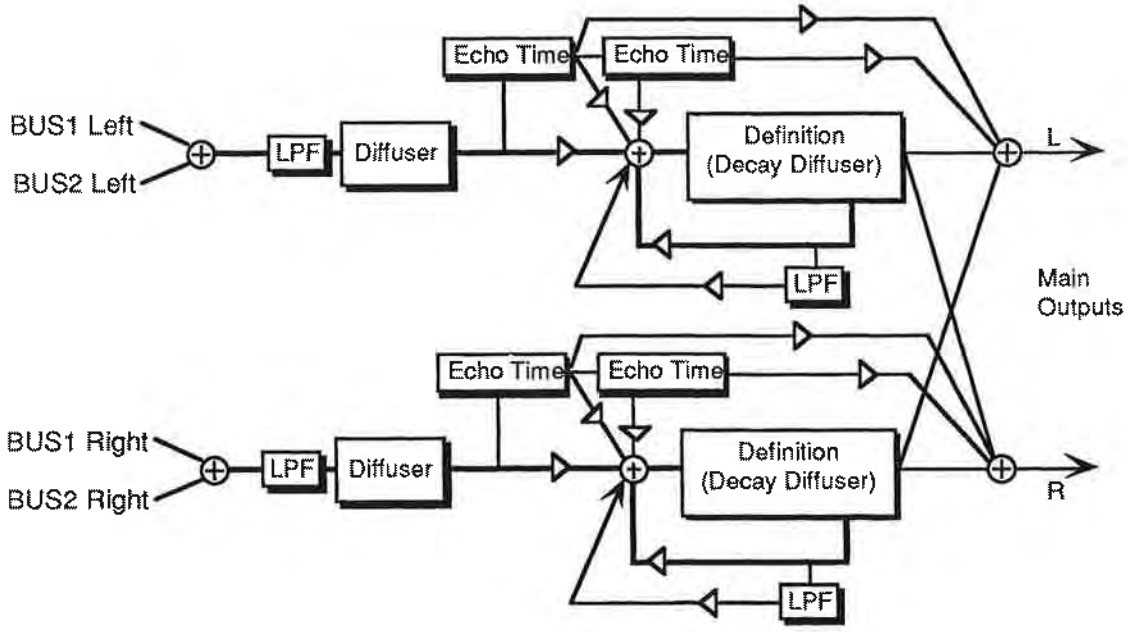
WAH RANGE
Range: -99 to +99
Controls how much depth the wah will have. This is analogous to how far you move a wah-wah pedal back and forth.

SYSTEM FEEDBACK
See the descriptions under CMP+DIST+REV.

ROM-14 SMALL ROOM
ROM-15 LARGE ROOM
ROM-16 HALL REVERB2

These three algorithms offer a studio quality reverb that can be controlled with a high degree of precision. The BUS1 and BUS2 parameters can be routed into the effect with different Dry/Wet mixes. BUS3 is a dry path. SMALL ROOM provides ambience; LARGE ROOM provides a larger ambience, and HALL REVERB2 is a large acoustic space providing a high density reverb.

SMALL ROOM, LARGE ROOM, HALL REVERB2 Signal Routing



These three reverb algorithms share essentially the same signal routing topology. The signal enters a low pass filter and goes directly through the diffusers which smear the signal. The signal is then routed to a larger decay diffuser, known as Definition, and is diffused over a period of time (creating a decay). There are taps from both the left and right Definition that are routed to the output to create a synthesized stereo output. A signal from the Definition goes through a low pass filter followed by a low frequency decay parameter, which controls the rate of decay of the low frequencies. There is also a parameter at this stage that controls the decay time of both the left and right signals. The left and right signals are routed back into the Definition. There are two echo times between the diffuser and the definition that can be routed directly to the output, or sent back through the definition. The BUS1 and BUS2 MIX parameters allow you to create a Dry/Wet mix. BUS3 (not shown) is a dry signal.

The parameters available for these three algorithms are:

DECAY TIME
Small Room Range: 0.20 to 100.0 sec.
Large Room Range: 0.20 to 150.0 sec.
Hall Reverb2 Range: 0.70 to 250.0 sec.
Controls the amount of time it takes for the reverberation to decay away to a very low level after the input signal stops. In the room reverbs we don't recommend higher settings, which tend to create an infinite and unnatural sustain. Since most rooms don't naturally have a large decay, set this low for the best sound. However, higher values are recommended for the hall reverb.

PREDELAY TIME
Range: 0 to 450 ms
Controls the amount of time it takes for the original signal to be presented to the reverb. Higher values denote a longer delay.

LF DECAY TIME
Range: -99 to +99
This parameter acts like a tone control and will boost (positive values) or cut (negative values) the rate at which low frequencies will decay.

HF DAMPING
Range: 0 to 99
This parameter controls the rate of attenuation of high frequencies in the decay of the reverberation. As natural reverb decays, some high frequencies tend to get absorbed by the environment. Increasing the value of this parameter will gradually filter out (dampen) more and more high frequency energy.

HF BANDWIDTH
Range: 1 to 99
The high frequency bandwidth acts as a low pass filter on the signal going into the reverb, controlling the amount of high frequencies that will pass into the effect. The higher the setting, the more high frequencies are allowed to pass. This functions like a tone control on a guitar.

DIFFUS1
Range: 0 to 99
This parameter smears the input signal transients, to *diffuse* and smooth the sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear (smoother sounding with fewer discrete echoes). We recommend settings of 50 for starters.

DIFFUS2
Range: 0 to 99
This parameter, similar to and in series with Room Diffusion1, performs the same way but controls lower frequency ranges. Experiment with different levels between the diffuser parameters to find the settings that are right for your source.

DECAY DEFINITION
Range: 0 to 99
This parameter controls the rate at which echo density is increased with time. Setting this parameter too high can cause the echo density to build at a rate which exceeds the decay rate. A general rule of thumb: DECAY DEFINITION should not exceed the LF DECAY TIME added to the DECAY TIME.

DETUNE RATE
Range: 0 to 99
This parameter controls the LFO rate of detuning introduced into the reverberation decay. Detuning creates a slight oscillating pitch shift into the decay, giving it a more natural sound by breaking up resonant modes.

DETUNE DEPTH
Range: 0 to 99
This parameter controls the depth of the detuning, that is, how much the pitch will change. Low values yield a metallic sound. Some sounds may require very low values, while others sound more natural with higher values.

PRIMARY SEND LEVEL
Range: -99 to +99
This parameter controls the level of the diffused input signal into the reverb definition.

REFL 1 TIME
Range: 0 to 120 milliseconds
This parameter controls the delay time for the first pre-echo. Pre-echoes are the first sounds that have been reflected back from the walls or reflective "live" surfaces. Higher values delay the diffused signal more.

REFL 1 LEVEL
Range: 0 to 99
This parameter controls the level of the first pre-echo. This pre-level controls the echo send to the Definition.

REFL 1 SEND
Range: 0 to 99
This parameter controls the level of the first pre-echo, with the echo routed directly to the output.

REFL 2 TIME
Range: 0 to 120 milliseconds
This parameter controls the delay time for the second pre-echo.

REFL 2 LEVEL
Range: 0 to 99
This parameter controls the level of the second pre-echo. As a signal continues to bounce off the different reflective surfaces (walls), it decreases in volume. This parameter should not be set too high, to create a natural sounding echo.

REF 2 SEND
Range: 0 to 99
This parameter controls the level of the second pre-echo, with the echo routed directly to the output.

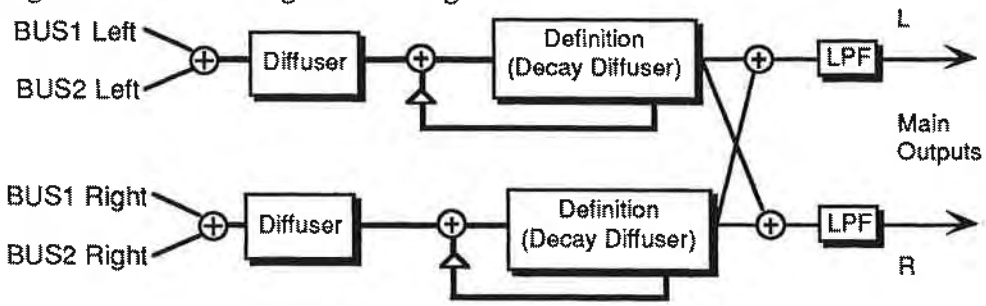
POSITN BAL (1)
POSITN BAL (2)
POSITN BAL (3)
Ranges: -99 to +99
The Position Balance parameters simulate the depth of the room/hall. Think of these parameters as three different microphones placed at various distances within the room/hall (the first is closest to the front, the third is farthest from the front). When the range (volume) is higher for POSITN BAL (1), the sound appears closer to the front, whereas a higher setting for POSITN BAL (3) appears farther from the front, suggesting a deeper (wetter) room/hall.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These modulation parameters are explained in detail in *Section 5 — Effect Concepts.*

ROM-17 SMALL PLATE
ROM-18 LARGE PLATE

A plate reverb takes the vibrations from a metal plate and uses them to create a metallic sounding reverb. SMALL PLATE is a tight sounding plate reverb, and LARGE PLATE simulates a larger plate reverb. Small plate reverbs are most often used in the studio for drums and percussion, while large plate reverbs are often used to enhance a vocalist's performance.

Small Plate, Large Plate Reverb Signal Routing



The two plate reverb algorithms share exactly the same signal routing topology. The internal values of the components (not user programmable) differentiate the large and small plate reverbs. The signal goes directly through the diffusers which smear the signal. The signal is then routed to a larger diffuser, known as Decay Definition, and is diffused over a period of time (creating a decay). The signal is then routed to the output, and then goes through a low pass filter. There is a parameter that controls the Decay Time of both the left and right signals (shown as triangles above). This signal is then routed back into the definition. There is also an external dry signal (not shown) that goes directly from the input to the output (BUS3). The BUS1 and BUS2 MIX parameters can be routed into the effect for different Dry/Wet mixes.

The user-controllable parameters for the plate reverbs algorithms are:

DECAY TIME
Small Plate Range: 0.2 to 100.0 sec.
Large Plate Range: 0.4 to 140.0 sec.
This parameter controls the amount of time it takes for the reverberation to decay. Percussion sounds best using SMALL PLATE. High values of decay time sound good on these algorithms.

PREDELAY TIME
Small Plate Range: 0 to 500 ms
Large Plate Range: 0 to 430 ms
This parameter controls the amount of time it takes for the input signal to be presented to the plate reverb. A value of 0 would offer no delay.

HF DAMPING
Range: 0 to 99
Increasing the value of this parameter will gradually filter out increasing amounts of high-frequency energy. Higher values yield an abrupt decay. This parameter controls the cut off of a low pass filter in series with the decay within the definition.

HF BANDWIDTH
Range: 1 to 99
This parameter acts as a low-pass filter on the output of the plate reverbs, controlling the amount of high frequencies present. The higher the setting, the more high frequencies are allowed to pass through, offering a brighter ringing sound. Some interesting effects can be created by using a mod controller over a large range.

DIFFUS1
Range: 0 to 99
The DIFFUS1 parameter smears the input signal to create a smoother sound. Lower values will cause impulse sounds to appear as a series of discrete echoes, while higher values tend to increase the smear, making the echoes less apparent.

DIFFUS2
Range: 0 to 99
This Diffuser, similar to and in series with the previous one, offers control over lower frequency ranges. Plate Reverbs tend to sound metallic, and the diffusers help to smear the signal, eliminating the metallic sound.

DECAY DEFINITION
Range: 0 to 99
This parameter controls the rate at which echo density increases with time. Higher values can cause the echo density to build at a rate that exceeds the decay rate. Try to select the highest value that works with your sound source for the best performance.

ER LEV (1 to 4)
Ranges: -99 to +99
These parameters control four early reflection levels. Setting these levels to lower values will produce a wetter sound. These four reflection levels are close to the input of the DECAY DEFINITION.

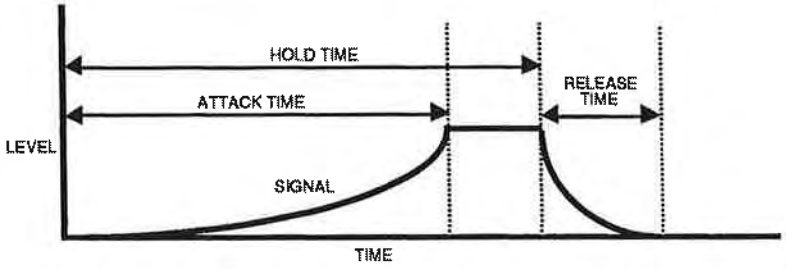
L/R BALANCE
Range: -99 to +99
This parameter controls the left/right stereo balance of the plate reverb signal. A setting of -99 would offer hard left, whereas a setting of +99 would offer hard right. A setting of +0 would place the reverb in the center of the stereo spectrum.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-19 REVRSE REVRB

REVRSE REVRB produces a reverberation that gradually increases, simulating a backwards sound with a maximum duration of several seconds. When a signal enters this algorithm, the plate reverb (from which this algorithm is derived) is almost instantaneously turned on, and then the output volume is ramped up. This algorithm will only trigger one time. REVERSE REVERB is triggered by an input signal level (THRESHOLD) determined by the user. Once triggered, the reverse envelope will proceed to completion, and ignore subsequent trigger levels. If you are looking for a reverse effect that will retrigger, try using REVRSE REVB 2. The topology of the Reverse Reverb is similar to the Plate Reverb. The BUS1 and BUS2 MIX parameters offer different dry/wet mixes.

The parameters available for this algorithm are:



HOLD TIME
Range: 1ms to 10.0s
This parameter controls the amount of time that the reverse envelope will sound after it has been triggered. Rule of thumb: don't set the HOLD TIME much longer than the ATTACK TIME.

ATTACK TIME
Range: 1ms to 10.0s
This parameter sets the duration over which the reverb builds. It is recommended that you set this value less than the HOLD TIME.

RELEASE TIME
Range: 1ms to 10.0s
This parameter determines the release time after the HOLD TIME has elapsed. Generally this time is very short. Lower values offer an abrupt cutoff.

TRIGGER THRESHOLD
Range: -96 to +0 dB
Set this parameter as low as possible to work with your particular sound source. To eliminate false triggering, it should not be set too low. When the input signal rises above this threshold, the reverse envelope will begin.

HF DAMPING
Range: 0 to 99
This parameter sounds best when set to low values. It has the same function as in the Plate Reverb, which is to filter out more and more high frequency energy. For the most natural sounding reverse effect, we recommend a setting of 0.

DIFFUS1
Range: 0 to 99
DIFFUS1 smears the input signal making a smoother sounding reverb. This parameter controls the high frequency ranges. For percussion sounds, high values are recommended.

DIFFUS2
Range: 0 to 99
Similar to and in series with DIFFUS1, this parameter controls lower frequency ranges.

DECAY DEFINITION
Range: 0 to 99
The DECAY DEFINITION controls the rate at which echo density is increased with time. If set too high, the echo density will build at a rate that exceeds the decay rate. This can be used for a special effect.

SLAPBACK TIME
Range: 0 to 530ms
This parameter controls the delay time of an internal dry signal to create a slapback. This effect helps to simulate a backwards reverb, since now the dry signal appears at the end. In general we recommend the BUS Mixes be set all wet (99) for this effect. Rule of thumb: Set this parameter at about the same value as the HOLD TIME.

SLAPBACK LEVEL
Range: 0 to 99
This parameter adjusts the volume of the slapback (internal dry) signal. A value of 0 would eliminate audible slapback.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-20 REVRSE REVB2

REVRSE REVB2 is identical to REVRSE REVRB, except this algorithm will retrigger by an assigned input signal level (threshold) determined by the user. Once triggered, the reverse envelope will proceed to completion, unless retriggered by subsequent input signals. If you are looking for a reverse effect that will not retrigger, try using the previous reverse reverb algorithm. The BUS1 and BUS2 MIX parameters are used to select different Dry/Wet mixes.

The parameters available for this algorithm are:

HOLD TIME
Range: 1ms to 10.0s
This parameter controls the amount of time that the reverse will sound after it has been triggered. Rule of thumb: Don't set the HOLD TIME much longer than the ATTACK TIME.

ATTACK TIME
Range: 1ms to 10.0s
This parameter sets the duration over which the reverb builds. It is recommended that you set this value less than the HOLD TIME.

RELEASE TIME
Range: 1ms to 10.0s
This parameter determines the RELEASE TIME after the HOLD TIME has elapsed. Generally this time is very short. Lower values offer an abrupt cutoff.

TRIGGER THRESHOLD
Range: -96 to +0 dB
Set this parameter as low as possible to work with your particular sound source. To eliminate false triggering, it should not be set too low. When the input signal rises above this threshold, the reverse envelope will begin.

PRE-TRIGGER TIME
Range: 0 to 530 ms
This parameter is used to capture transients that occur before the trigger. This parameter is critical to the sound quality. The user determines how much pretrigger sound will be injected into the reverse reverb tank.

HF DAMPING
Range: 0 to 99
This parameter sounds best when set to low values. Its function is to filter out more and more high frequency energy. For the most natural sounding reverse effect, we recommend a setting of 0.

DIFFUS1
Range: 0 to 99
DIFFUS1 smears the input signal making a smoother sounding reverb. This parameter controls the high frequency ranges. For percussion sounds, high values are recommended.

DIFFUS2
Range: 0 to 99
Similar to and in series with DIFFUS1, this parameter controls lower frequency ranges.

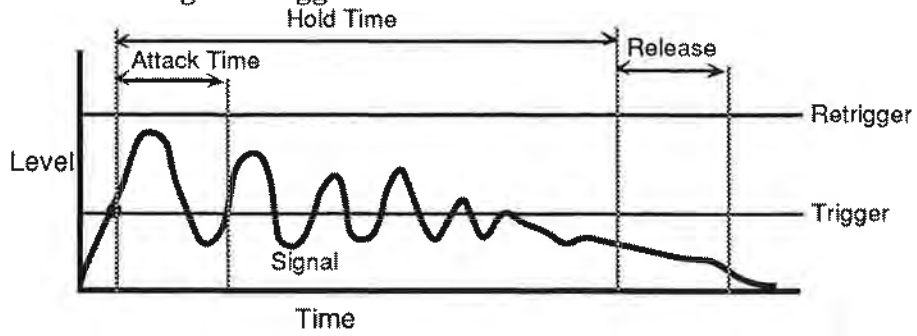
DECAY DEFINITION
Range: 0 to 99
DECAY DEFINITION controls the rate at which echo density is increased with time. If set too high, the echo density will build at a rate that exceeds the decay rate. This can be used for a special effect.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

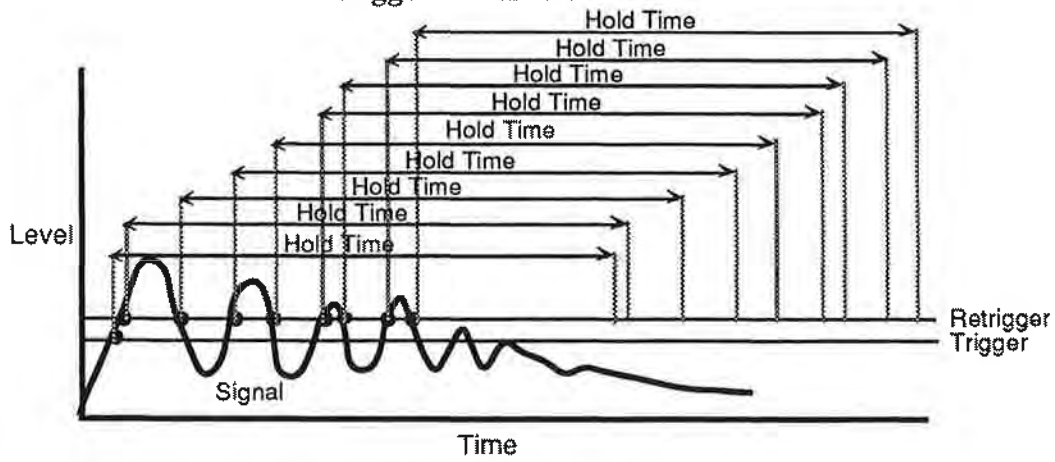
ROM-21 GATED REVERB

GATED REVERB provides an excellent gated reverb. When a reverb is muted part way through, it creates a gated sound. To achieve this gated effect, both the Gated and Reverse reverbs must gate a number of internal parameters, not just the output amplitude envelope. It is however, the output amplitude over which the user has control. The ASR–10 offers a highly controllable gated reverb, optimized for percussive instruments, but useful for any input signal. The gate is first opened when the input signal passes the trigger threshold. This trigger threshold is set as low as possible by the user, so that none of the input signal is missed. The gated reverb is distinguished from the reverse reverb by retriggering whenever the input signal passes a retrigger (user programmable) threshold (see diagrams). The gate will stay open as long as the input signal remains above the retrigger threshold, and all the input signals will be accumulated under this gate until the total input signal level falls below the retrigger threshold. When this happens, the Hold Time will begin (as shown in the diagram below). The reason there are two thresholds is to eliminate false retriggering and to ensure precise hold time durations. If you desire a separate gate on each note, use the Non Lin reverbs. The topology for the Gated Reverb is derived from the Plate Reverb. The BUS1 and BUS2 parameters can be routed into the effect with different Dry/Wet mixes.

Gated Reverb with a High Retrigger Threshold



Gated Reverb with a Low Retrigger Threshold



The parameters available for the Gated Reverb algorithm are:

ATTACK TIME
Range: 1ms to 10.0s
This parameter sets the ATTACK TIME of the gated reverb once the incoming signal has reached the trigger level. Generally the attack should be short and not set longer than the HOLD TIME. This parameter should not be used to achieve a reverse reverb envelope, because here the attack volume increases whereas in the Reverse Reverb the attack volume accelerates.

HOLD TIME
Range: 1ms to 10.0s
This parameter sets the amount of time that the reverb will hold after the retrigger and before the release. The HOLD TIME will begin again if retriggered.

DECAY TIME
Range: 0.2 to 100.0 sec.
This parameter controls the amount of time it takes for the reverberation to decay.

RELEASE TIME
Range: 1ms to 10.0s
This parameter sets the amount of time after the HOLD TIME has elapsed for the gated reverb to shut down. Generally these times are very short.

TRIGGER THRESH
Range: -96 to +0 dB
This parameter sets the signal level that triggers the gated reverb. When the incoming signal reaches this value, it triggers (starts) the gated reverb. Higher values would require a stronger incoming signal. Set this parameter as low as possible to work with your particular source, but not too low to cause false triggering.

RETRIGGER THRESH
Range: -96 to +0 dB
This parameter sets the level at which the gated reverb will retrigger. For precise HOLD TIME that begins at the onset of the incoming source, this parameter should be set higher than the incoming signal to prevent retriggering. After the incoming signal reaches the trigger threshold, the gated reverb is activated. Every time the signal reaches the retrigger threshold, the gated reverb will retrigger causing the HOLD TIME to restart.

If the level of this parameter is set lower than the incoming signal, the gated reverb will continue to retrigger. With a high DECAY TIME, this adds a cavernous quality to percussion instruments.

HF DAMPING
Range: 0 to 99
This parameter sounds best when set to low values. Its function is to filter out more and more high frequency energy. For the most natural sounding reverse effect, we recommend a setting of 0.

DIFFUS1
Range: 0 to 99
DIFFUS1 smears the input signal making a smoother sounding reverb. This parameter controls the high frequency ranges. For percussion sounds, high values are recommended.

DIFFUS2
Range: 0 to 99
Similar to and in series with DIFFUS1, this parameter controls lower frequency ranges.

DEFINITION
Range: 0 to 99
DEFINITION controls the rate at which echo density is increased with time. If set too high, the echo density will build at a rate that exceeds the decay rate. This can be used for a special effect.

SLAPBACK
Range: 0 to 500 ms
This parameter controls the delay time of an internal dry stereo signal to create a slapback. In general the SLAPBACK is greater or equal to the HOLD TIME to achieve a reverse effect.

SLAPBACK LEVEL
Range: 0 to 99
This parameter adjusts the volume of the slapback (internal dry) signal. A value of 0 would eliminate any audible slapback.

ER LEV (1 to 4)
Ranges: -99 to +99
These parameters control four Early Reflection Levels. Setting these levels to lower values will produce a wetter sound.

L/R BALANCE
Range: -99 to +99
This parameter controls the left/right stereo balance of the gated reverb signal. A setting of -99 would offer hard left, whereas a setting of +99 would offer hard right. A setting of +0 would place the reverb in the center of the stereo spectrum.

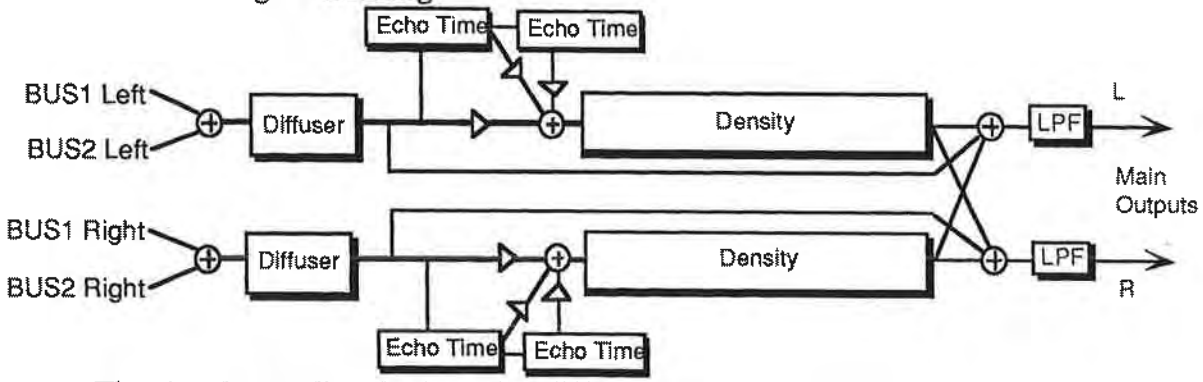
MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-22 NLIN REVRB 1
ROM-23 NLIN REVRB 2
ROM-24 NLIN REVRB 3

Non Lin Reverbs can be used to obtain blooming reverb, gated reverb, reverse reverb and early reflections. In general, they do not produce an exponentially decaying reverb. Unlike the Hall, Room and Plate reverbs, NLIN REVRB 1, 2, and 3 pass the input signal through the reverb diffusers only once. For this reason the reverb diffusers are called *Density*, to distinguish them from the other reverb diffusers (called Definition). Density controls the *amount* of echo density, as opposed to the rate of increase of echo density. Other reverbs give limited control of early reflections. The Non Lin reverbs purposely impose a coloration on the resulting sound. The BUS1 and BUS2 parameters can be routed into the effect with different Dry/Wet mixes.

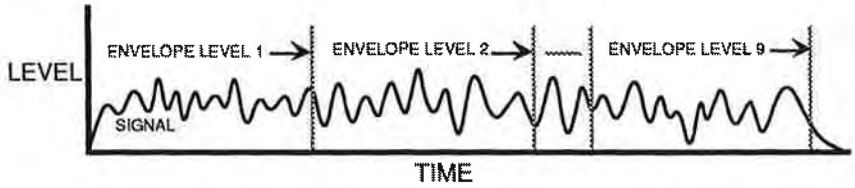
- NLIN REVRB 1 is optimized for shorter duration effects (approx. 0.5 sec.)
- NLIN REVRB 2 offers approx. a 1.5 sec. duration
- NLIN REVRB 3 is sonically similar to NLIN REVRB 1, but there is less stereo movement, making it better suited for drum tracks

Non Lin Reverb Signal Routing



The signal goes directly through a diffuser which smears the signal. The signal is then routed to a decay diffuser, known as Density, and is diffused over a period of time. Within the density the signal goes through a high-frequency damper. The signal is then routed to the output. After the density, the signal passes through a low pass filter. There are two echo times between the diffuser and the density. There is also an external dry signal (not shown) that goes directly from the input to the output (BUS3). The BUS1 and BUS2 MIX parameters can be routed into the effect with different Dry/Wet mixes.

The parameters available for the Non Lin Reverbs are:



ENV LEV (1 to 9)
Ranges: 0 to 99
These parameters control the output tap levels sequenced in time across the density from input to output. Envelope Level 1 is tapped right after the diffusers and before the echoes (see the diagram). If this is undesirable, set ENV LEV1 to 0. Envelope Levels 8 and 9 are positioned at the very end of the Density; setting these too high can cause excessive ringing. Envelope Levels 8 and 9 are also very dry. Set all nine tap levels to find the envelope for your application. We recommend the average Envelope Level not to exceed a value of 45 to prevent overdriving these reverbs.

HF DAMPING
Range: 0 to 99
HF DAMPING is located within the density. This parameter selects the amount of high frequency energy to be filtered out.

HF BANDWIDTH
Range: 1 to 99
The high frequency bandwidth parameter acts as a low pass filter on the output signal, controlling the amount of high frequencies that will be heard. The higher the setting the more high frequencies are heard. This works the same way that a tone control would work on a guitar.

DIFFUS1
Range: 0 to 99
This parameter smears the input signal transients of higher frequency ranges. Higher values are recommended for smoother percussion. Very low values will give a highly repetitive echo-like sound. DIFFUS1 and 2 exist within each diffuser block (see diagram).

DIFFUS2
Range: 0 to 99
DIFFUS2 is similar to DIFFUS1, but offers control of lower frequencies. In general a setting of 50 can be considered an equal mix of dry/diffused sound; this setting is a good starting point.

DENSTY 1
Range: 0 to 99
Density 1 controls the number of echoes.

DENSTY 2
Range: 0 to 99
Density 2 controls the number of echoes in a lower frequency range. In general to get the smoothest sound, DENSTY 2 is usually less than the value of DENSTY 1.

PRIMARY SEND LEVEL
Range: -99 to +99
This parameter controls the level of the diffused input signal which is nearly instantaneous with respect to the input. This signal is injected directly into the Density at the specified level.

REFL 1 TIME
Non Lin 1, 3 Range: 0 to 600 ms
Non Lin 2 Range: 0 to 85 ms
This controls the amount of time it takes for the first pre-echo to be injected into the Density. Pre-echoes are the sounds that have been reflected back from the walls or other reflective surfaces.

REFL 1 SEND LEVEL
Range: -99 to +99
This parameter controls the level of the first pre-echo.

REFL 2 TIME
Non Lin 1, 3 Range: 0 to 600 ms
Non Lin 2 Range: 0 to 85 ms
This controls the amount of time it takes for the second pre-echo to be injected into the Density.

REFL 2 SEND LEVEL
Range: -99 to +99
This parameter controls the level of the second pre-echo. Experiment with both positive and negative on all echoes to change the tonal character of the results.

L/R BALANCE
Range: -99 to +99
This parameter controls the left/right stereo balance of the gated reverb signal. A setting of -99 would offer hard left, whereas a setting of +99 would offer hard right. A setting of +0 would place the reverb in the center of the stereo spectrum.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-25 MULTITAP DDL

MULTITAP DDL produces four independent controllable delays. The BUS1, BUS2, and BUS3 MIX parameters can be routed into the effect with different dry/wet mixes. This algorithm sounds best with a mix of wet and dry.

DLY 1 TIME
Range: 0 to 2186 ms
This parameter sets the amount of delay time for the independent delays. Experiment with different settings to find the right mix for your sound source and application. Some interesting effects can be implemented by using a real time modulation controller for these parameters.

DLY 1 RGN
Range: 0 to 99
This parameter determines the amount of signal that will be fed from the output back into the input, increasing the number of repeats in the delays. A setting of 99 would offer an infinite delay.

DLY 1 LEVEL
Range: 0 to 99
This parameter adjusts the volume of the delayed signals against the original dry signal. A level of 0 will offer no audible delay.

DLY 1 PAN
Range: -99 to +99
This parameter determines the location of the delays in the stereo spectrum. A value of -99 is panned far left, and +99 is far right.

DLY 2 TIME
DLY 2 RGN
DLY 2 LEVEL
DLY 2 PAN
DLY 3 TIME
DLY 3 RGN
DLY 3 LEVEL
DLY 3 PAN
DLY 4 TIME
DLY 4 RGN
DLY 4 LEVEL
DLY 4 PAN
Each of the four discrete delays has identical parameters and ranges as described above.

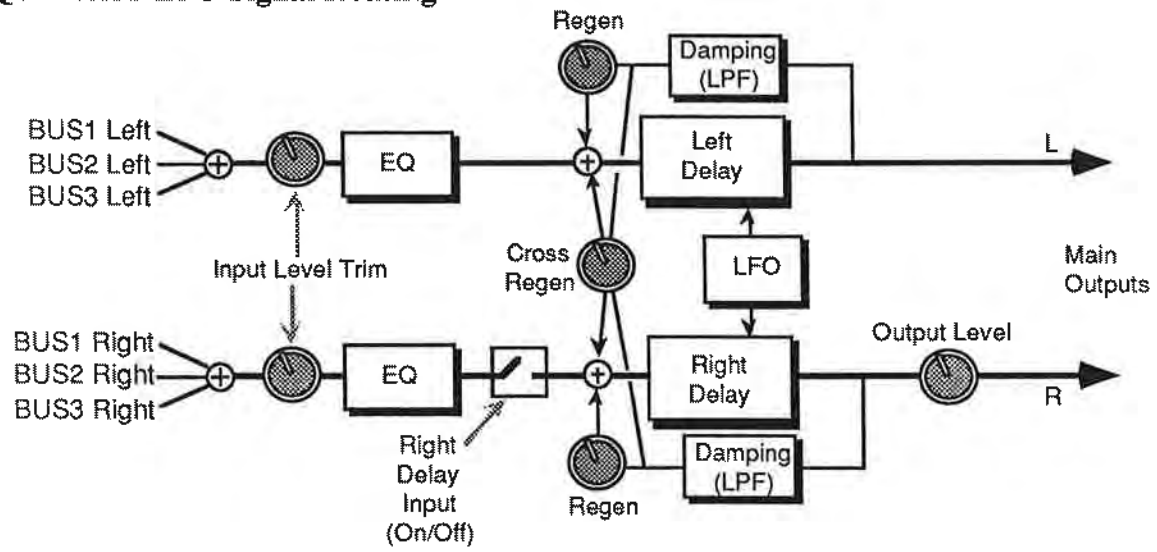
REGEN DAMPING
Ranges: 0 to 99
This parameter controls the cutoff of a low pass filter on the feedback signal, which adjusts the amount of damping to the feedback signals. The higher the number, the more the signals are dampened.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-26 EQ + DELAY LFO

EQ + DELAY LFO features a stereo digital delay that provides LFO modulation of a wide range of delays. This algorithm sounds great with an electric piano, but try it with any source!

EQ+DELAY LFO Signal Routing



The BUS1, BUS2, and BUS3 MIX parameters can be routed into the effect with different Dry/Wet mixes.

The parameters available for this algorithm are:

DLY TIME L
Range: 0 to 845 ms
This parameter determines the amount of time between the input signal and the left delay output.

DLY TIME R
Range: 0 to 845 ms
This parameter determines the amount of time between the input signal and the right delay output. Set this differently from the left delay time to achieve dotted 1/8th note type effects.

LFO RATE
Range: 0 to 99
This parameter controls the rate of pitch modulation which is the LFO. To achieve a chorusing effect, this rate must be very slow.

LFO WIDTH
Range: 0 to 99
This parameter controls the excursion of pitch modulation. Since the rate is usually very slow, then the width is usually large.

L/R LFO
Range: OUT-OF-PHASE or IN-PHASE
When this parameter is IN-PHASE, the left and right choruses will modulate their detunes together. When set to OUT-OF-PHASE, the detune on the left channel will go up while the detune on the right will go down.

DELAY REGEN
Range: -99 to +99
This parameter controls the amount of regen (regeneration) applied to the delay time taps. The sign of the value determines the polarity of the regen.

DLY CROSS REGEN
Range: -99 to +99
This parameter allows you to feedback the delayed signals to their opposite sides; the left voice crosses to the right voice, and the right voice crosses to the left voice. A setting of +99 or -99 will cause infinite delay. Be careful, if the delay regen is set too high, it may cause this parameter to "blow up."

REGEN DAMPING
Range: 0 to 99
This parameter adjusts the cutoff of a low pass filter on the feedback signal, which controls the amount of damping to the feedback signals. The higher the number, the more the signals are damped.

R DELAY INPUT
Range: OFF or ON
This parameter disables the input into the right side delay line. The right delay line will still get input from the DDL CROSS REGEN. This allows a ping-pong delay effect.

R OUTPUT LEVEL
Range: 0 to 99
This parameter controls the right output signal level.

BASS FC
Range: 0 to 1000 Hz
This parameter sets the cutoff frequency of the lower frequency band shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the low shelving filter.

TREBLE FC
Range: 1KHz to 15KHz
This parameter sets the cutoff frequency of the high shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the high shelving filter.

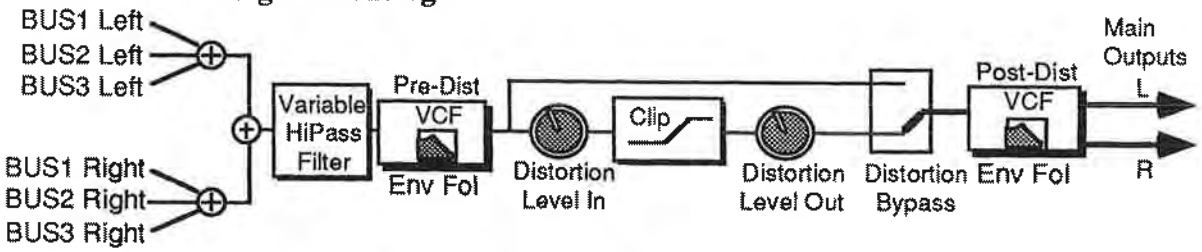
EQ TRIM
Range: -24 to +0 dB
This parameter allows you to adjust the input volume of the EQs to eliminate the possibility of clipping boosted signals.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-27 VCF+DISTORT

VCF+DISTORT combines a voltage control filter and a raspy distortion. Three effects can be obtained: Distortion, Wah-wah, and Auto-wah. The last two functions use the same VCF. These filters can be disabled or used as EQ if desired. There is a second VCF that exists after the distortion. It can be set to act like a simple speaker simulator, or it can be modulated in parallel with the pre-distortion VCF.

VCF+DISTORT Signal Routing



The BUS1, BUS2, and BUS3 MIX parameters can be routed into the effect with different Dry/Wet mixes.

The parameters available for this algorithm are:

DIST LEVEL IN
Range: 0 to 99
This parameter controls the gain going into the distortion effect. DIST LEVEL IN will boost the signal level up to 48 dB. For more distortion, use a high input level gain and turn the DIST LEVEL OUT down to keep the volume under control. For less distortion, use a low gain input level and a higher output volume.

DIST LEVEL OUT
Range: 0 to 99
This parameter controls the gain coming out of the distortion effect. Generally, if the DIST LEVEL IN is set high, set this parameter lower to control the volume.

PRE-DIST VCF FC
Range: 1 to 99
This parameter determines the filter cut off frequency before the distortion. Higher values have a brighter sound. This parameter can be modulated, using a CV Pedal for a wah-wah pedal effect. To disable the distortion filter, set this parameter to 99. To use as an EQ, set the desired value and make sure PRE-DIST ENV FOL is +0. To use as the auto-wah, set this parameter close to 1 (lower values) and turn on PRE-DIST ENV FOL.

PRE-DIST VCF Q
Range: 1 to 25
This parameter determines the level and width of the resonant peak at the filter cutoff point. While the PRE-DIST VCF FC parameter determines where (at what frequency) this peak will occur, the PRE-DIST VCF Q setting controls the *presence* of the peak. This setting is important for the auto-wah effect.

PRE-DIST ENV FOL
Range: -99 to +99
This parameter determines how much the amplitude of the incoming signal will modify the distortion filter cutoff frequency. When set to +0, no modification will occur. When set to mid positive values, the PRE-DIST VCF FC will go high, but then come down to its nominal setting. When set to negative mid values, the PRE-DIST VCF FC will go low, and then go back up to its nominal setting How quickly it does so is determined by the ATTACK and RELEASE parameters. This sound is the auto-wah; positive values will boost the high frequencies, offering an "oww-oww" sound, and negative values will cut the high frequencies, producing a "dweep-dweep" sound.

POST-DIST VCF FC
Range: 01 to 99
This parameter determines the filter cut off frequency *after* the distortion. Higher values have a brighter sound. This parameter can also be modulated, using a CV Pedal for a wah-wah pedal effect. To disable the post-distortion filter, set this parameter to 99.

POST-DIST VCF Q
Range: 1 to 25
This parameter determines the level and width of the resonant peak at the filter cutoff point. While the FC (filter cutoff) parameter determines where (at what frequency) this peak will occur, the Q setting controls the presence of the peak. This setting is important for the auto-wah effect.

POST-DIST ENV FOL
Range: -99 to +99
This parameter determines how much the amplitude of the incoming signal will modify the distortion filter cutoff frequency. When set to +0, no modification will occur. When set to mid positive values, FC will go high, but then come down to its nominal setting. When set to negative mid values, FC will go low, and then go back up to its nominal setting How quickly it does so is determined by the ATTACK and RELEASE parameters. This sound is the auto-wah; positive values will boost the high frequencies, offering an "oww-oww" sound, and negative values will cut the high frequencies, producing a "dweep-dweep" sound.

ATTACK
Range: 50µs to 10.0s
This parameter sets the attack of the envelope follower (i.e. determines how closely the attack is followed) once the incoming signal has been detected. Generally the attack should be short.

RELEASE
Range: 1ms to 10.0s
This parameter sets the amount of time after the incoming signal has ceased for the envelope follower to shut down. Generally these times are longer than the attack times.

DISTORTION BYPASS
Range: OFF or ON
This parameter allows you to bypass the distortion (as shown on the signal routing diagram).

PRE-EQ HP FC
Range: 0 to 1000 Hz
This parameter filters out the low frequencies before the EQ. The higher the value, the less low frequencies will pass through.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-28 GUITAR AMP 1
ROM-29 GUITAR AMP 2

These algorithms recreate the warm sound of a guitar amplifier. They do this by emulating tube distortion characteristics. These algorithms are good for all stringed instruments. GUITAR AMP 1, designed for Hard Rock, offers more distortion than GUITAR AMP 2, which is optimized for "bluesy" type sounds. The BUS1 and BUS2 MIX parameters can be routed into the effect with different Dry/Wet mixes.

The parameters available for this algorithm are:

PREAMP GAIN
Range: -48 to +48 dB
This parameter adjusts the amount of boost or cut applied to the incoming signal. We recommend a setting of +0 dB, since these emulations were optimized for distortion there. Lower preamp gains will result in less distortion, while higher preamp gains will yield clipping distortion. For low preamp gain, it may be desirable to use low tube bias values.

OUTPUT LEVEL
Range: 0 to 99
This parameter controls the output level of the main amp before the output EQ.

TUBE BIAS
Range: 0 to 99
For preamp gains approximately 0 dB, this parameter controls the emphasis of even to odd harmonics which determines the tone of the amp; mid values emphasize even harmonics and offer a warmer ("glowing tube") sound, while the highest values may sound like tubes going bad. Tube bias and preamp gain are independent parameters. For low preamp gain, it may be desirable to use low tube bias values, because this more closely imitates the operation of a real amplifier.

PRE-EQ TRIM
Range: -24 to +0 dB
This parameter controls the input level to the pre-amp EQ to eliminate the possibility of clipping boosted signals.

PRE-EQ HIGH
Range: 4 to 1000 Hz
This parameter filters out the low frequencies before the preamp. The higher the value, the less low frequencies pass through.

PRE-EQ FC
Range: 0 to 9999 Hz
This parameter determines the center frequency of the parametric filter before the preamp. Higher values have a brighter sound.

PRE-EQ GAIN
Range: -48 to +24 dB
This parameter adjusts the amount of boost or cut applied to the parametric filter in front of the preamp.

PRE-EQ Q
Range: 1 to 18
This parameter determines the width of the resonant peak at the parametric filter center frequency. While the Filter center parameter determines where (at what frequency) this peak will occur, the Q setting controls the presence of the peak.

GATE OFF
Range: -96 to +0 dB
This parameter sets the lower threshold level at which the noise gate shuts off the audio.

GATE ON
Range: -96 to +0 dB
This parameter sets the upper threshold level at which the noise gate passes the audio. The higher second threshold prevents false "turn ons."

GATE RELEASE
Range: 1ms to 10.0s
This parameter sets the amount of time after the signal has elapsed for the noise gate to shut down. For a longer sustain, set this parameter higher.

HIGH PASS FC
Range: 4 to 1000 Hz
This parameter filters out the low frequencies of the main amp prior to the speaker. The higher the value, the less low frequencies pass through.

EQ1 FC
Range: 0 to 9999 Hz
This parameter determines the filter center frequency of the parametric in the main amp stage. Higher values have a brighter sound.

GAIN
Range: -48 to +24 dB
This parameter adjusts the amount of boost or cut applied to the main amp parametric.

EQ1 Q
Range: 1 to 18
This parameter determines the width of the resonant peak of the filter center. While the Filter center parameter determines where (at what frequency) this peak will occur, the Q setting controls the presence of the peak.

EQ2 FC
Range: 0 to 9999 Hz
This parameter determines the filter center frequency of the second parametric in the main amp stage. Higher values have a brighter sound.

GAIN
Range: -48 to +24 dB
This parameter adjusts the amount of boost or cut applied to the second main amp parametric.

EQ2 Q
Range: 1 to 18
This parameter determines the width of the resonant peak of the second filter center.

LOW PASS FC
Range: 2.0 to 15.0 KHz
This parameter filters out the high frequencies of the speaker. The lower the value, the less high frequencies pass through. This speaker filter is less selective than the speaker cabinet emulation algorithms.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-30 GUITAR AMP 3

GUITAR AMP 3 combines an inverse expander with a bright distortion for amp lead sounds. The inverse expander may be thought of as a compressor that amplifies all signals below the threshold. This algorithm is good for heavy metal and hard rock guitar solos. The BUS1 and BUS2 parameters can be routed into the effect with different Dry/Wet mixes.

The parameters available for this algorithm are:

PREAMP GAIN
Range: -48 to +48 dB
This parameter adjusts the amount of boost or cut applied to the EQ'd incoming signal. Lead sounds are obtained using high gain.

OUTPUT LEVEL
Range: 0 to 99
This parameter controls the output level before the output EQ.

PRE-EQ TRIM
Range: -24 to +00 dB
This parameter controls the input level to the preamp EQ to eliminate the possibility of clipping boosted signals.

PRE-EQ FC
Range: 0 to 9999 Hz
This parameter determines the filter center frequency of the parametric in the preamp stage. Higher values have a brighter sound.

PRE-EQ GAIN
Range: -48 to +24 dB
This parameter adjusts the amount of boost or cut applied to the preamp parametric.

PRE-EQ Q
Range: 1 to 18
This parameter determines the width of the resonant peak at the filter center. While the Filter center parameter determines where (at what frequency) this peak will occur, the Q setting controls the presence of the peak.

EXPANSION RATIO
Range: 1:1 to 40:1, infinity
This parameter sets the amount of inverse expansion. Expansion occurs below the threshold. If this is set to 3:1 for example, it will expand the change in signals below the threshold by three times in an attempt to make the signal amplitude approach the threshold level.

THRESHOLD
Range: -96 to +0 dB
This parameter sets the inverse expander threshold level. Signals beneath this level will be expanded, while signals that are above will be unaffected. As the input signal dies away below the threshold, the expander will increase the gain of the signal.

GATE OFF
Range: -96 to +0 dB
This parameter sets the lower threshold level at which the noise gate shuts off the audio.

GATE ON
Range: -96 to +0 dB
This parameter sets the upper threshold level at which the noise gate passes audio. This higher second threshold prevents false "turn ons."

GATE RELEASE
Range: 1ms to 10.0s
This parameter sets the amount of time after the signal has elapsed for the noise gate to shut down. For a longer sustain, set this parameter higher.

HIGH PASS FC
Range: 4 to 1000 Hz
This parameter filters out the low frequencies of the main amp prior to the speaker. The higher the value, the less low frequencies pass through.

EQ1 FC
Range: 0 to 9999 Hz
This parameter determines the filter center frequency of the parametric in the main amp stage. Higher values have a brighter sound.

GAIN
Range: -48 to +24 dB
This parameter adjusts the amount of boost or cut applied to the main amp parametric.

EQ1 Q
Range: 1 to 18
This parameter determines the width of the resonant peak at the filter center. While the Filter center parameter determines where (at what frequency) this peak will occur, the Q setting controls the presence of the peak.

EQ2 FC
Range: 0 to 9999 Hz
This parameter determines the filter center frequency of the second parametric in the main amp stage. Higher values have a brighter sound.

GAIN
Range: -48 to +24 dB
This parameter adjusts the amount of boost or cut applied to the second main amp parametric.

EQ2 Q
Range: 1 to 18
This parameter determines the width of the resonant peak of the second filter center.

LOW PASS FC
Range: 2.0 to 15.0 KHz
This parameter filters out the high frequencies of the speaker. The lower the value, the less high frequencies pass through. True speaker emulations are provided as separate algorithms.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-31 SPKR CABINET

SPKR CABINET simulates the warm sound of an opened back speaker cabinet. Speaker Cabinet is fabulous for a guitar, bass or any other stringed instrument. This algorithm contains the resonances and the nonlinearity of a real musical instrument speaker. The BUS1 and BUS2 parameters can be routed into the effect with different Dry/Wet mixes. For a brighter speaker emulation, try using TUNABLE SPKR.

The parameters available for this algorithm are:

OUTPUT GAIN
Range: -48 to +24 dB
Since speaker cabinets are "lossy," output gain is required to compensate losses in perceived volume. Setting this gain too high will cause clipping of the output signal.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-32 TUNABLE SPKR

Tunable Speaker offers an EQ controllable speaker sound that is brighter than SPKR CABINET. By tuning three parametric filters, you can simulate many different speaker cabinet sounds that are used in all styles of music. The BUS1 and BUS2 parameters can be routed into the effect with different Dry/Wet mixes. The parameters available for this algorithm are:

MID1 FC
Range: 0 to 9999 Hz
This parameter sets the center of the mid-frequency parametric. Higher values have a brighter sound.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of cut (negative values) or boost (positive values) applied to this mid-frequency parametric.

MID1 Q
Range: 1 to 18
This parameter is a bandwidth control that determines the width of the resonant peak at the center of the frequency band. By raising the value, you can produce a narrower bandwidth.

MID2 FC
GAIN
MID2 Q
MID3 FC
GAIN
MID3 Q
These parameters are identical to the previous ones, but can be assigned to control different bandwidths within the mid-range.

INPUT TRIM
Range: -24 to +0 dB
This parameter allows you to adjust the input level before the EQs to eliminate the possibility of clipping boosted signals.

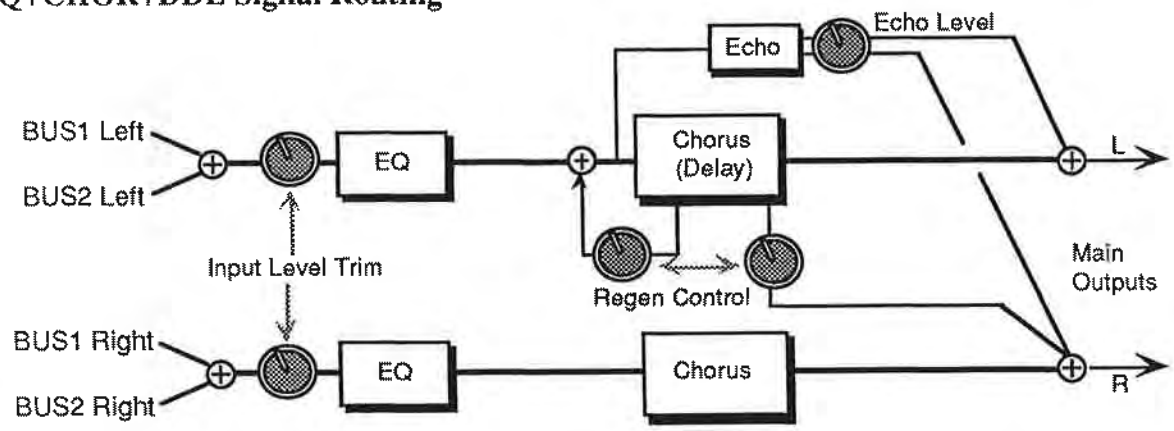
OUTPUT GAIN
Range: -48 to +24 dB
Since speaker cabinets are "lossy," output gain is required to compensate losses in perceived volume. Setting this gain too high will cause clipping of the output signal.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-33 EQ+CHOR+DDL

EQ+CHOR+DDL combines an EQ with a chorus and a digital delay. This is the industry standard chorus effect, designed with very long delays to provide a modulated detune effect. This algorithm sounds great with a guitar, but try it with any source!

EQ+CHOR+DDL Signal Routing



The signal enters a programmable EQ, which is preceded by an input level trim (EQ TRIM). The signal is then routed to the chorus which is heard directly at the output. There is also a delayed unchorused signal (sharing the same delay lines) that is routed back into the chorus. There is also a second signal from the delay line that is routed to the right side. There are two discrete echo times tapped out before the chorus delay line. These are unchorused echoes. The signal from the echoes is routed directly to the outputs. There is also a dry signal (not shown) that goes directly from the input to the output (BUS3). The BUS1 and BUS2 MIX parameters can be routed into the effect with different Dry/Wet mixes.

The parameters available for the EQ+CHOR+DDL are:

CHOR RATE
Range: 0 to 99
This parameter controls the rate of pitch modulation which is the chorus. To achieve chorusing, this rate must be very slow.

CHOR WIDTH
Range: 0 to 99
This parameter controls the excursion of pitch modulation. Since the rate is usually very slow, then the width is usually very large.

CHORUS CENTER
Range: 0 to 99
This parameter controls the nominal delay time of the chorus about which the delay modulation occurs. Adjusting this parameter will change the tonal character of the effect. This delay time is not related to the regen delays or the echo delays.

L/R LFO
Range: OUT-OF-PHASE or IN-PHASE
When this parameter is IN-PHASE, the left and right choruses will modulate their detunes together. When set to OUT-OF-PHASE, the detune on the left channel will go up while the detune on the right will go down.

DLY TIME L
Range: 0 to 1500 ms
This parameter controls the time delay for the left channel regen delay, and has nothing to do with the chorus effect.

DLY TIME R
Range: 0 to 1500 ms
This parameter controls the time delay for the right channel regen delay, and has nothing to do with the chorus effect.

DELAY REGEN
Range: -99 to +99
This parameter controls the amount of regeneration applied to the delay time taps. The sign of the value determines the polarity of the regen.

ECHOTIME L
Range: 0 to 1500 ms
This parameter controls the chorus echo time for the left side. Higher settings yield a deeper echo. There are two discrete echoes, one to the left and one to the right.

ECHOTIME R
Range: 0 to 1500 ms
This parameter controls the chorus echo time for the right side. Higher settings yield a deeper echo.

ECHO LEVEL
Range: 0 to 99
This parameter controls the volume of the discrete echo for both the left and right sides. Higher values offer louder echo, while a value of 0 will eliminate the echo. For sustained sounds, mid echo levels yield a "poor man's reverb."

BASS FC
Range: 0 to 1000 Hz
This parameter sets the cutoff frequency of the lower frequency band shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the low shelving filter.

TREBLE FC
Range: 01KHz to 15KHz
This parameter sets the cutoff frequency of the high shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the high shelving filter.

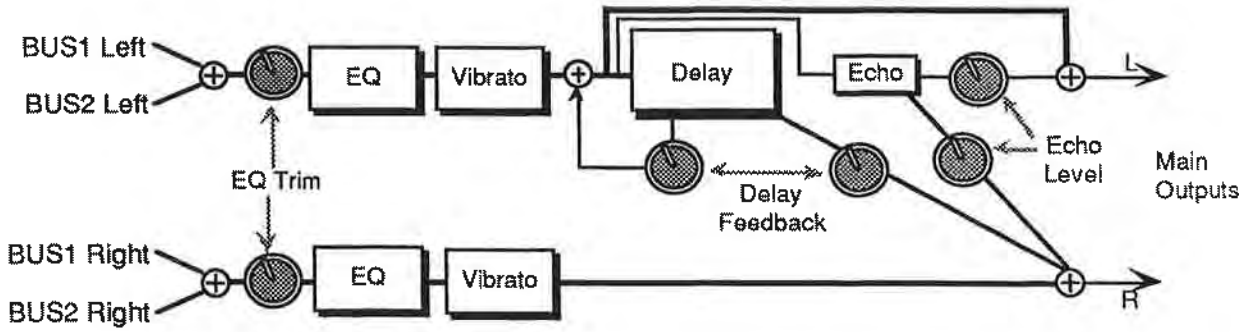
EQ TRIM
Range: -24 to +00 dB
This parameter allows you to adjust the input volume of the EQs to eliminate the possibility of clipping boosted signals.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-34 EQ+VIBR+DDL

EQ+VIBR+DDL combines a vibrato effect (a pitch shifter modulating over a very small range) with EQ and digital delay. Many vintage guitar amplifiers offered a vibrato control, but don't feel that this effect is limited to guitars, try this with other sources as well. There is a sample & hold parameter that doesn't hold the instantaneous pitch shift, but if set properly will provide a "chirping" effect when acting on the input signal.

EQ+VIBR+DDL Signal Routing



In this effect the signal enters a programmable EQ, which is preceded by an input level trim. The signal is then routed to the vibrato. The vibrato is routed directly to the output. The vibrato also passes through the delay which is then regenerated back into the delay. A different delay signal is routed to the output on the right side. This constitutes a "ping-pong" delay effect. The Delay Feedback parameter between the delay sends controls the delay feedback amount. There are also two echoes that are sent to the left and right outputs respectively. The signal from the two echoes has one level control. There is also an external dry signal (not shown) that goes directly from the input to the output (BUS3). The BUS1 and BUS2 MIX parameters can be routed into the effect with different Dry/Wet mixes.

The parameters available for this algorithm are:

VIBR RATE
Range: 0 to 99
This parameter sets the amount of modulation. Higher values create a faster vibrating rate.

WIDTH
Range: 0 to 99
This parameter sets the width (adjusts the amount) of the modulation.

L/R LFO
Range: OUT-OF-PHASE or IN-PHASE
This parameter controls the vibrato pitch direction of the left and right channels. When OUT-OF-PHASE, the quadrature pitch change on the left channel will lag 90° from the right. When IN-PHASE, both channels will change pitch together.

SAMPLE/HOLD RATE
Range: 0 to 100
This parameter controls the sample rate of a sample and hold network. This is applied to the LFO within the vibrato. When in hold (low values), it causes rhythmic chirps in the pitch of the audio signal. Higher values will increase the number of holds per second, making the vibrato flow more smoothly. The sample and hold function is turned off when set to 0.

DELAY TIME L
Range: 0 to 1500 ms
This parameter controls the time delay on the left regenerated delay.

DELAY TIME R
Range: 0 to 1500 ms
This parameter controls the time delay on the right nonregenerated delay.

DELAY FEEDBACK
Range: -99 to +99
This parameter controls the amount of positive or negative feedback applied to the regenerated delay. The sign of the value determines the polarity of the feedback. A value of +0 will eliminate any feedback. This parameter controls both left and right levels.

ECHOTIME L
Range: 0 to 1500 ms
This parameter controls the echo time for the left side. Higher settings yield a deeper echo. There are two discrete echoes, one to the left and one to the right.

ECHOTIME R
Range: 0 to 1500 ms
This parameter controls the echo time for the right side.

ECHO LEVEL
Range: 0 to 99
This parameter controls the volume of the discrete echo for both the left and right sides. A setting of 0 will eliminate any audible echo.

BASS FC
Range: 0 to 1000 Hz
This parameter selects the cutoff frequency of the low shelving filter.

BASS GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the low shelving filter.

TREBLE FC
Range: 01KHz to 15KHz
This parameter selects the cutoff of the upper frequency band high shelving filter.

TREBLE GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the high shelving filter.

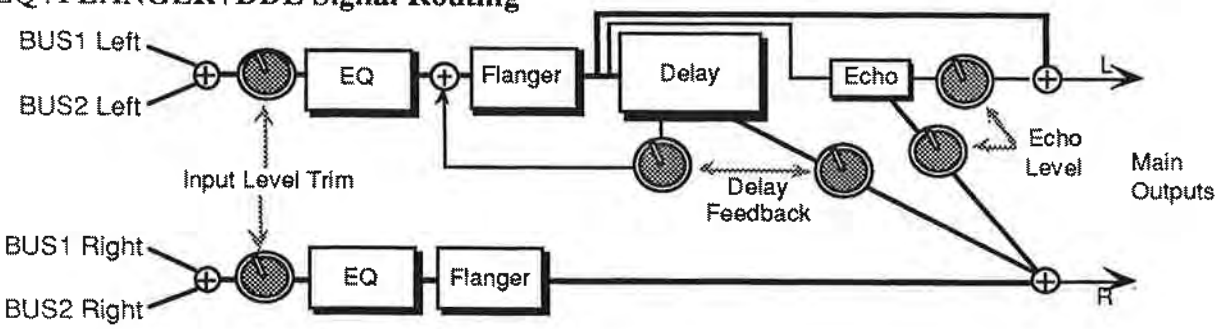
EQ TRIM
Range: -24 to +0 dB
This parameter allows you to adjust the input volume before the EQs to eliminate the possibility of clipping boosted signals.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-35 EQ+FLNGR+DDL

EQ+FLNGR+DDL combines an EQ with a flanger and a digital delay. Use flanging to get that "jet aircraft woosh" effect.

EQ+FLANGER+DDL Signal Routing



In this algorithm the signal enters an input level trim (EQ TRIM) followed by a programmable EQ, and then is routed to the flanger. The flanger is routed directly to the output. The left channel signal passes through the delay and is routed back into the flanger. Another signal from the delay is routed to the output on the right side. The DELAY FEEDBACK parameter controls both delay levels. There are two discrete echoes that are sent to the left and right outputs respectively. The signal from both echoes has one level control (ECHO LEVEL). There is also an external dry signal (not shown) that goes directly from the input to the output (BUS3). The BUS1 and BUS2 MIX parameters can be routed into the effect with different Dry/Wet mixes.

The parameters available for this algorithm are:

FLANGE RATE
Range: 0 to 99
This parameter controls the rate of modulation of the flanger effect.

WIDTH
Range: 0 to 99
This parameter controls the range of the high to low frequency sweep in the flanger effect.

FLANGE CENTER
Range: 0 to 99
This parameter controls the sweep center of the flanger effect. The larger the flanger center, the wider will be the available width.

FLANGE FEEDBACK
Range: -99 to +99
This parameter controls the amount of feedback applied from the output to the flanger input. The sign of the value determines the polarity of the feedback.

NOTCH DEPTH
Range: -99 to +99
This parameter controls the depth of the notches created by the flanging effect. A setting of +0 will disable the flanging effect. A setting of +0 will also provide a doppler effect for wide moderately slow LFO rates.

L/R LFO
Range: OUT-OF-PHASE or IN-PHASE
This parameter determines whether the flanger on the left and right channels is modulating IN-PHASE or OUT-OF-PHASE.

SAMPLE/HOLD RATE
Range: 0 to 100
This parameter controls the sample rate of a sample and hold network. This is applied to the LFO within the flanger. When in hold, the effect will be to create momentarily fixed notches within the frequency spectrum (if the notch depth is not +0). A setting of 1 will have the largest space between samples. Higher values will increase the number of holds per second, making the flanging flow more smoothly. The sample and hold function can be turned off (when set to 0).

DLY TIME L
Range: 0 to 1500 ms
This parameter controls the time delay for the left channel regen delay. This is the "ping."

DLY TIME R
Range: 0 to 1500 ms
This parameter controls the time delay for the right channel regen delay. This is the "pong."

DELAY FEEDBACK
Range: -99 to +99
This parameter controls the level of the delay time taps. The sign of the value determines the polarity of the feedback.

ECHO TIME L
Range: 0 to 1500 ms
This parameter controls the flanger echo time for the left side. Higher values yield a deeper echo.

ECHO TIME R
Range: 0 to 1500 ms
This parameter controls the flanger echo time for the right side.

ECHO LEVEL
Range: 0 to 99
This parameter controls the volume of the discrete echoes for both the left and right sides. A setting of 0 would eliminate any audible echo.

BASS FC
Range: 0 to 1000 Hz
This parameter sets the cutoff frequency of the lower frequency band shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the low shelving filter.

TREBLE FC
Range: 1KHz to 15KHz
This parameter selects the cutoff of the upper frequency band high shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the high shelving filter.

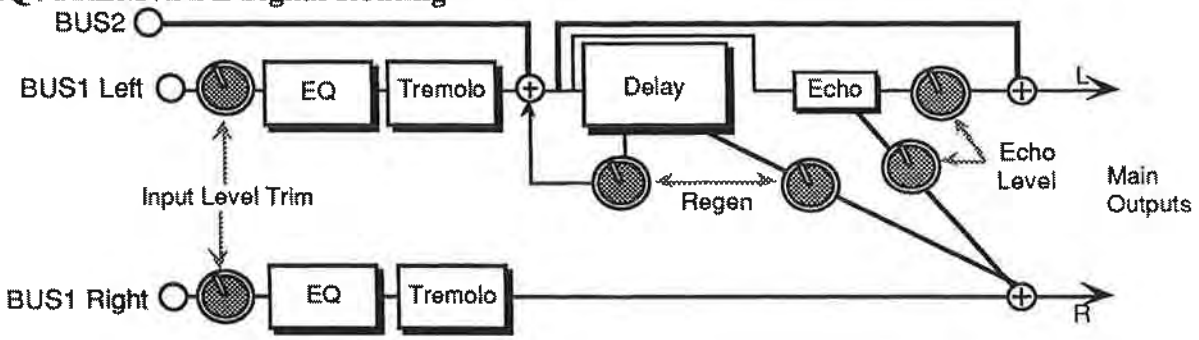
EQ TRIM
Range: -24 to +0 dB
This parameter allows you to adjust the input volume of the EQs to eliminate the possibility of clipping boosted signals.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-36 EQ+TREM+DDL

EQ+TREM+DDL combines a tremolo effect, which is a pulsating change in volume, with an EQ and a digital delay.

EQ+TREM+DDL Signal Routing



The signal enters an input level trim (EQ TRIM) followed by a programmable EQ, and is then routed to the tremolo. The tremolo is routed directly to the output. The left channel signal passes through the digital delay and is routed back into the delay. There is another signal from the delay that is routed to the output on the right side. One regen parameter (DELAY FEEDBACK) between the delay sends controls the left and right delay level. This constitutes the "ping-pong" effect. There are two echoes that are sent to the left and right outputs respectively. The signal from the two discrete echoes has one level control. There is also an external dry signal (not shown) that goes directly from the input to the output (BUS3). The parameters available for this algorithm are:

BUS1 DLY AFT TREM — Delay After Tremolo
Range: 0 to 99
This parameter determines how much of the signal that is sent to BUS 1 (the tremolo) will also go to the delay — it is a delay mix for the tremolo signal.

BUS2 DELAY MIX
Range: 0 to 99
This parameter controls the Dry/Wet mix of the delay for voices sent to BUS2. There is no tremolo routed to BUS2 (see diagram).

TREM RATE
Range: 0 to 99
This parameter sets the rate of modulation. Mid values create a faster wavering sound. This parameter, when used in conjunction with the SAMPLE/HOLD RATE parameter, can create some interesting staccato effects.

DEPTH
Range: 0 to 99
This parameter sets the depth of amplitude modulation.

L/R LFO
Range: OUT-OF-PHASE or IN-PHASE
This parameter controls whether the left and right channels of the stereo tremolo will modulate IN-PHASE or OUT-OF-PHASE.

SAMPLE/HOLD RATE
Range: 0 to 100
This parameter controls the sample rate of a sample and hold network applied to the LFO within the tremolo. When in "Hold," the effect will be to fix the instantaneous amplitude (if the depth is not 0). A setting of 1 will have the largest space between holds. Lower settings create a staccato effect, whereas higher values will increase the amount of samples, making the tremolo flow more smoothly. This parameter can also be turned off (when set to 0).

DLY TIME L
Range: 0 to 1500 ms
This parameter controls the time delay for the left channel regen delay, and has nothing to do with the tremolo effect.

DLY TIME R
Range: 0 to 1500 ms
This parameter controls the time delay for the right channel regen delay.

DELAY FEEDBACK
Range: -99 to +99
This parameter controls the amount of regen applied to the delay time taps. The sign of the value determines the polarity of the regen. A value of +0 will eliminate the audible delay.

ECHO TIME L
Range: 0 to 1500 ms
This parameter controls the tremolo echo time for the left side. Higher settings yield a deeper echo.

ECHO TIME R
Range: 0 to 1500 ms
This parameter controls the tremolo echo time for the right side.

ECHO LEVEL
Range: 0 to 99
This parameter controls the volume of the discrete echo for both the left and right sides.

BASS FC
Range: 0 to 1000 Hz
This parameter selects the cutoff frequency of the lower frequency band shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the low shelving filter.

TREBLE FC
Range: 1KHz to 15KHz
This parameter selects the cutoff frequency of the upper frequency band high shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the high shelving filter.

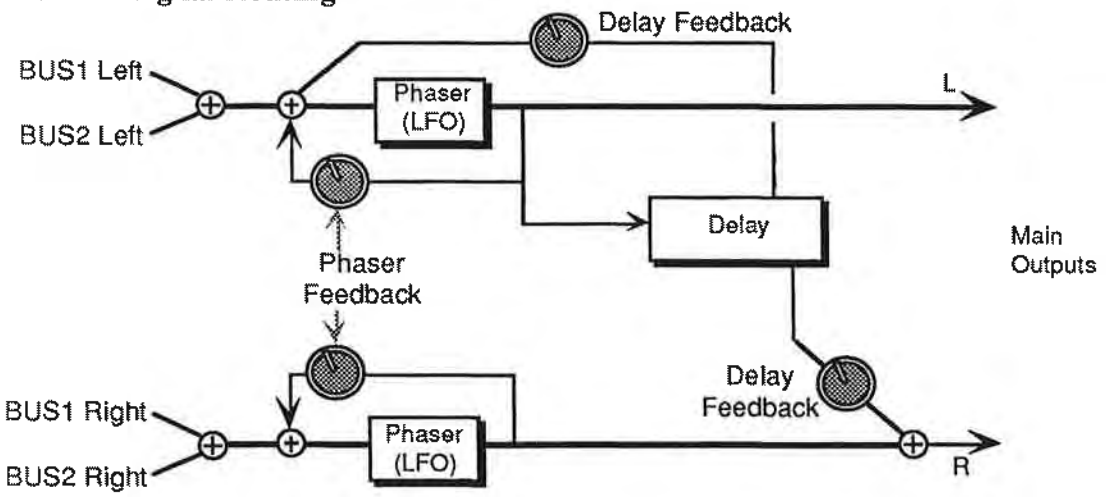
EQ TRIM
Range: -24 to +0 dB
This parameter allows you to adjust the input volume of the EQs to eliminate the possibility of clipping boosted signals.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-37 PHASER+DDL

PHASER+DDL combines a phaser with a digital delay. The phaser creates non-harmonically spaced moving notches in the signal spectrum, whereas a flanger creates harmonic spacing. This phaser implements a stereo twelve pole phasing network to achieve time delay which is a function of frequency; this is what differentiates the phaser from the flanger. The phasing effect is achieved within the Phaser topology, so it does not depend upon the external mix. A delay is included at the left output of the phaser that feeds back into the phaser (see the diagram). Setting the phaser DELAY FEEDBACK parameter (shown at the knobs) to 0 will disable this delay function. The DELAY FEEDBACK also controls the delay feed forward level of another tap sent to the right channel. This delay topology achieves a 1.5 second ping-pong effect, and is very effective as a "poor man's reverb." The BUS1 and BUS2 parameters can be routed into the effect with different Dry/Wet mixes. BUS3 (not shown) is a dry path.

Phaser+DDL Signal Routing



The parameters available for this algorithm are:

PHASE RATE
Range: 0 to 99
The LFO is within the phaser network. This parameter controls the rate of modulation of the phaser poles. The higher the value the faster the rate. Lower values work best with sustained signals.

WIDTH
Range: 0 to 99
This parameter controls the width of the notch excursion. For large excursions set this parameter to 99 which can give a very high "woosh" and a very low "woosh."

PHASE CENTER
Range: -99 to +99
This parameter controls the phaser pole center. High values raise the nominal spectral location of the "woosh" sound, while low values lower the "woosh." The range from high to low is controlled with the phaser width.

PHASER FEEDBACK
Range: -99 to +99
This parameter controls the amount of feedback applied to the left and right channel phaser. The sign of the value determines the polarity of the feedback.

NOTCH DEPTH
Range: -99 to +99
This parameter controls the depth of the notches created by the phasing. Deep notches occur in the phased spectrum when the parameter is set to 99. When this parameter is set to +0, there exists no phasing (i.e. notches), but there is a doppler effect with higher phase rates.

L/R LFO
Range: OUT-OF-PHASE or IN-PHASE
This parameter determines whether the phaser on the left and right channels is modulating IN-PHASE or OUT-OF-PHASE.

SAMPLE/HOLD RATE
Range: 0 to 100
This parameter controls the sample rate of a sample and hold network applied to the LFO within the phaser. When in hold, the effect will be to create momentarily fixed notches within the frequency spectrum (if the notch depth is not +0). A setting of 1 will have the largest space between samples. Higher values will increase the number of holds per second, making the phasing flow more smoothly. The sample and hold function can be turned off (when set to 0).

DLY TIME L
Range: 0 to 1600 ms
This parameter sets the delay time for the left side. This is the "ping."

DLY TIME R
Range: 0 to 1600 ms
This parameter controls the feed forward delay time for the right side. This is the "pong."

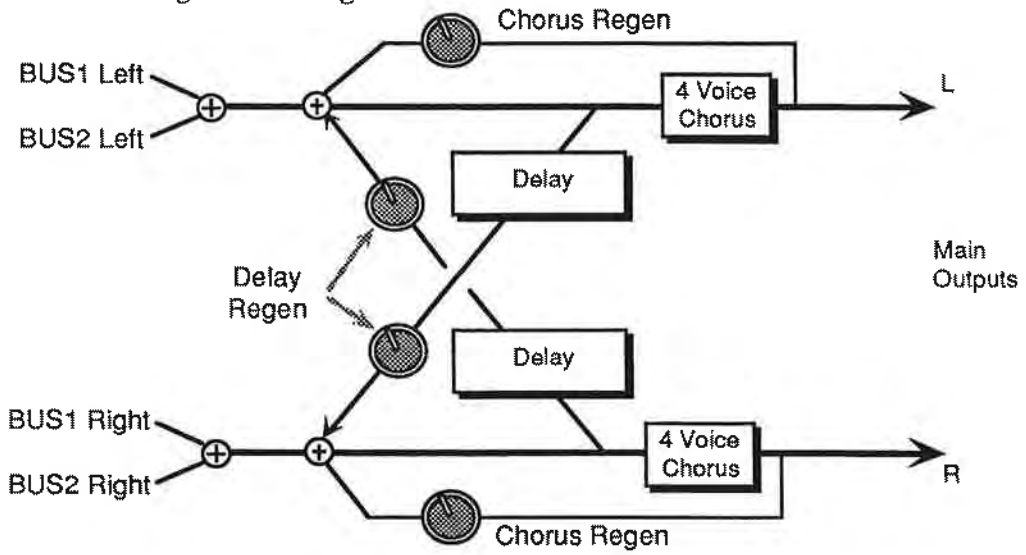
DELAY FEEDBACK
Range: -99 to +99
This parameter controls the feedback amount for the delay effect. The sign of the value determines the polarity of the feedback. A value of +0 will eliminate the delay effect. This parameter also controls the feed forward level.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-38 8-VOICE CHOR

8-VOICE CHOR offers a symphonic chorused sound having eight different voices and using eight separately randomized LFOs. This algorithm also offers a user programmable stereo delay in a cross coupled configuration between the left and right chorused outputs (see diagram). This algorithm is good for creating an ensemble of instruments from single sources (there is no internal filtering applied to any of the chorused voices). The BUS1 and BUS2 parameters can be routed into the effect with different Dry/Wet mixes. BUS3 (not shown) is a dry path. For this algorithm we recommend a Mix of around 50 for a starting point.

8-VOICE CHOR Signal Routing



LFO RATE
Range: 0 to 99
This parameter controls the eight ganged rates of modulation of the respective voices. This modulation produces an effect similar to both vibrato and tremolando occurring at the same time.

WIDTH
Range: 0 to 99
This parameter controls the excursion of the vibrato of all the individual voices.

STEREO SPREAD
Range: 0 to 99
This parameter offers a synthesized stereo field. The highest value is true stereo, intermediate values have the left and right signals mixed on both sides, and the lowest value yields only the left input channel from the right and left outputs. This parameter, though not a stereo pan, provides some interesting stereo effects when controlled by a modulation source.

CHORUS REGEN
Range: 0 to 99
This parameter determines the amount of signal that will be fed from the output of the chorus back into the input of the chorus. A value of 0 will eliminate the regeneration effect.

L REGEN TIME
Range: 0 to 800 ms
This parameter controls the amount of time that the non-chorused signal will delay for the left channel.

R REGEN TIME
Range: 0 to 800 ms
This parameter controls the amount of time that the non-chorused signal will delay for the right channel.

DELAY REGEN
Range: 0 to 99
This parameter determines the amount of signal that will be fed from the delay output back into the chorus input, increasing the number of repeats in the delay for high values. A value of 0 will eliminate the delay effect.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-39 PITCH SHIFT

The Pitch Shifters allow you to change the pitch of a signal to any pitch desired within a range of one octave in either direction. The ASR–10 offers three different pitch shift algorithms, each designed with a different purpose:

- PITCH SHIFT offers a 1 unit splicer type pitch shifter.
- PITCH+DDL combines a pitch shifter with a DDL (described later).
- FAST PITCHSH is a 1 unit pitch shifter designed for pitch correction.

These algorithms sound best with a Mix of wet and dry. Try using a modulation controller for the Mix parameter to bring in or fade out the pitch shifted signal. Try the different pitch shifters until you find the one that works best with your sound source, and for your application. This first algorithm is a "splicer type" of pitch shifter, which means that it drops or adds small sections of the original signal to the effect. PITCH SHIFT does not incorporate zero crossing detection. This pitch shifter is best used for a doubling effect. Splicer type pitch shifters are popular because for low pitch shift ratios, splicing is infrequent. These pitch shifters can create very interesting stereo fields by panning each of two pitch shifted voices selectively, and because of the inherent time delay modulation of the algorithm. The BUS1 and BUS2 parameters can be routed into the effect with different Dry/Wet mixes. BUS3 (not shown) is a dry path.

The parameters available for this algorithm are:

VC 1 SEMI
Range: -12 to +12
This parameter allows you to adjust the pitch of Voice 1 up to an octave above or below the original pitch in semi-tones (half steps).

VC 1 FINE
Range: -99 to +99
This parameter allows you to fine tune the pitch of Voice 1.

VC 1 VOL
Range: 0 to 99
This parameter allows you to adjust the volume of Voice 1. A setting of 0 would eliminate any audible pitch shift.

VC 1 PAN
Range: -99 to +99
This parameter allows you to assign the location of Voice 1 in the stereo field. A value of -99 would be far left, and +99 would be far right.

VC 2 SEMI
Range: -12 to +12
This parameter allows you to adjust the pitch of Voice 2 up to an octave above or below the original pitch in semi-tones.

VC 2 FINE
Range: -99 to +99
This parameter allows you to fine tune the pitch of Voice 2.

VC 2 VOL
Range: 0 to 99
This parameter allows you to adjust the volume of Voice 2. A setting of 0 would eliminate any audible pitch shift.

VC 2 PAN
Range: -99 to +99
This parameter allows you to assign the location of Voice 2 in the stereo field. A value of -99 would be far left, and +99 would be far right.

QUALITY
Range: LONG/SMOOTHER or SHORT/COARSER
This parameter allows you to choose between a long/smooth setting, or a short/coarse setting. A smooth setting would sound best with slower sustaining chords, whereas a coarse setting would enhance a rapidly played musical passage. Depending on your sound source and musical needs, set this parameter accordingly. This parameter actually controls the effect transport delay; smooth yields a long transport delay, coarse yields a short transport delay.

LFO RATE
Range: 0 to 99
This parameter controls the rate of pitch modulation that creates a chorusing effect. To achieve chorusing, this rate must be very low.

WIDTH
Range: 0 to 99
This parameter controls the excursion of pitch modulation. Since the rate is usually very low, then the width is usually very large.

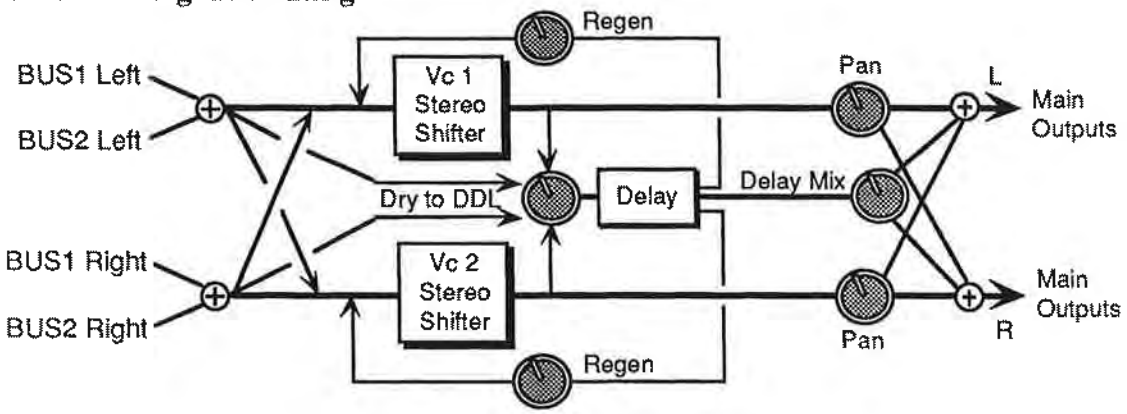
MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

Note: When KBD is selected as a MOD SRC, and VC1 SEMI or VC2 SEMI is selected as the DST (destination), the pitch will increase one semitone at a time above middle C (C4), and decrease one semitone at a time below middle C (Middle C being zero). After each octave, the pitch shift will "wrap" and start again (the pitch can't be modulated more than one octave in either direction).

ROM-40 PITCH+DDL

PITCH+DDL combines a pitch shifter with a digital delay. PITCH+DDL uses a continual crossfading technique of pitch shifting. This technique maintains the stereo field exactly. This pitch shifter works best for large pitch shift ratios, in some circumstances. Another feature of this algorithm is a digital delay that feeds back into the pitch shift. This feature allows spiraling upward or downward pitch shifts. The BUS1 and BUS2 parameters can be routed into the effect with different Dry/Wet mixes. BUS3 (not shown) is a dry path. This algorithm sounds best with a Mix of wet and dry. Try using a modulation controller for this parameter to fade in or fade out the pitch shifted signal.

PITCH+DDL Signal Routing



The parameters available for this algorithm are:

VC 1 SEMI
Range: -12 to +12
This parameter allows you to adjust the pitch of Voice 1 up to an octave above or below the original pitch in semi-tones.

VC 1 FINE
Range: -99 to +99
This parameter allows you to fine tune the pitch of Voice 1.

VOICE 1 VOL
Range: 0 to 99
This parameter adjusts the volume of Voice 1.

VOICE 1 PAN
Range: -99 to +99
This parameter allows you to assign the location of Voice 1 in the stereo field. A value of -99 would be far left, and +99 would be far right.

VC 2 SEMI
Range: -12 to +12
This parameter allows you to adjust the pitch of Voice 2 up to an octave above or below the original pitch in semi-tones.

VC 2 FINE
Range: -99 to +99
This parameter allows you to fine tune the pitch of Voice 2. Slight shifts create something like a chorused effect.

VOICE 2 VOL
Range: 0 to 99
This parameter adjusts the volume of Voice 2.

VOICE 2 PAN
Range: -99 to +99
This parameter allows you to assign the location of Voice 2 in the stereo field. A value of -99 would be far left, and +99 would be far right.

DRY LEVEL TO DDL
Range: 0 to 99
This parameter allows you to bypass the pitch shifter with an internal dry signal, and send it through the digital delay. Higher values would send more of the dry signal to the delay. The purpose of this parameter is to mix the dry signal appropriately with the pitch shifted delay signals.

DLY TIME L
Range: 0 to 1500 ms
This parameter controls the amount of time for the pitch shifted signal to delay from the left input.

DLY TIME R
Range: 0 to 1500 ms
This parameter controls the amount of time for the pitch shifted signal to delay from the right input.

DLY MIX
Range: 0 to 99
This parameter controls the mix between the delay signal and the pitch shifted signal. A setting of 0 would be all pitch shifter, and no delay. A setting of 99 will be all delay and no direct pitch shift.

DLY REGEN
Range: -99 to +99
This parameter controls the amount of feedback from the output of the delay back into the input of the pitch shifter. This allows you to create special effects with ascending/descending delays.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-41 FAST PITCHSH

FAST PITCHSH has a transport delay of only 10 ms and a maximum detune ratio of one semitone. Try shifting the voices slightly in both positive and negative values to create a fat sound. This algorithm should be used for pitch correction (for instance, try using the mod wheel as a pitch controller). The BUS1 and BUS2 parameters can be routed into the effect with different Dry/Wet mixes. BUS3 is a dry path. Depending on its usage, this algorithm generally sounds best with a Mix of dry and wet. Try using the modulation wheel for the Mix parameter to bring in or fade out the pitch shifted signal.

The parameters available for this algorithm are:

VOICE 1 FINE
Range: -99 to +99
This parameter allows you to fine tune the pitch of Voice 1.

VOICE 1 VOL
Range: 0 to 99
This parameter allows you to adjust the volume of Voice 1. A setting of 0 would eliminate any audible pitch shift.

VOICE 1 PAN
Range: -99 to +99
This parameter allows you to assign the location of Voice 1 in the stereo field. A value of -99 would be far left, and +99 would be far right.

VOICE 2 FINE
Range: -99 to +99
This parameter allows you to fine tune the pitch of Voice 2.

VOICE 2 VOL
Range: 0 to 99
This parameter allows you to adjust the volume of Voice 2. A setting of 0 would eliminate any audible pitch shift.

VOICE 2 PAN
Range: -99 to +99
This parameter allows you to assign the location of Voice 2 in the stereo field. A value of -99 would be far left, and +99 would be far right.

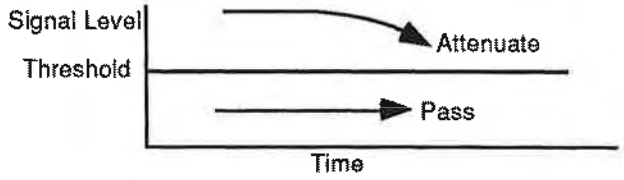
LFO RATE
Range: 0 to 99
This parameter controls the rate of pitch modulation that creates a chorusing effect. To achieve chorusing, this rate must be very low.

WIDTH
Range: 0 to 99
This parameter controls the excursion of pitch modulation. Since the rate is usually very low, then the width is usually very large.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-42 EQ+CMPRESSOR

EQ+CMPRESSOR combines an EQ with a full feature compressor. For high compressor ratios, this algorithm functions as a limiter. This algorithm operates by compressing (attenuating) signals above the threshold and passing the signals below the threshold. For higher ratios and lower thresholds, this algorithm can be used to create sustain. EQ exists in both signal and side chain paths, in contrast to the EXPANDER that has filtering in only the side chain path.



The parameters available for this algorithm are:

COMP GAIN
Range: -48 to +48 dB
This parameter boosts the compressed signal level.

COMP RATIO
Range: 1/1 to 40/1, infinity
This parameter sets the amount of compression. The range is based on decibels (dB) above the threshold. If this is set to 4/1 for example, it will compress changes in signals above the threshold by one quarter. When this is set to infinity, it acts as a limiter.

COMP THRESHOLD
Range: -96 to +0 dB
This parameter sets the threshold level. Signals that exceed this level will be compressed, while signals that are below will be unaffected. To turn off the compressor, set the level to +0 dB.

COMP ATTACK
Range: 50µs to 100ms
This parameter determines the attack rate after the initial signal has been detected and before the compression takes affect.

COMP RELEASE
Range: 1ms to 10.0s
This parameter determines how long it takes for the compression to be fully deactivated after the input signal drops below the threshold level. This is generally chosen longer than the attack time.

GATE OFF
Range: -96 to +0 dB
This parameter sets the lower threshold level at which the noise gate shuts off the audio.

GATE ON
Range: -96 to +0 dB
This parameter sets the upper threshold level at which the noise gate passes audio. This higher second threshold prevents false "turn ons."

GATE RELEASE
Range: 1ms to 10.0s
This parameter determines how long it takes for the gate to be fully released after the input signal drops below the threshold level. Lower settings yield a quick gate.

BASS FC
Range: 0 to 1000 Hz
This parameter sets the cutoff frequency of the lower frequency band shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the low shelving filter.

TREBLE FC
Range: 1KHz to 15KHz
This parameter sets the cutoff frequency of the upper frequency band high shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the high shelving filter.

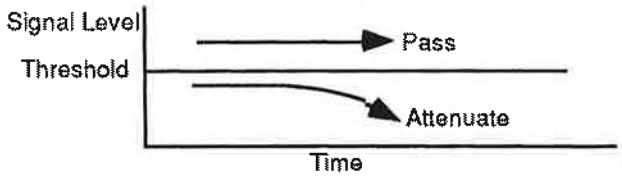
EQ TRIM
Range: -24 to +0 dB
This parameter allows you to adjust the input volume of the EQs, to eliminate the possibility of clipping boosted signals.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-43 EXPANDER

EXPANDER performs downward expansion of input signals. For high expansion ratios this algorithm functions as a gate. This algorithm operates by expanding (attenuating) signals below the threshold and passing the signals above the threshold. The Threshold is a parameter defined by the user. This algorithm can be used to eliminate noise. There is no EQ in the audio path; high and low pass filtering is provided on the side chain only. This algorithm possesses two unique features:

1) The ADSR in this algorithm has Attack, Sustain, and Release (the sustain is called the HOLD TIME).
2) This algorithm contains a trigger mask function. This function is used primarily to extract a click track from drum tracks. Once triggered, this function inserts a zero signal level into the side chain detector for an amount of time determined by the user. This function becomes triggered if TRIGGER MASK is enabled and if the side chain signal falls below the Trigger MASK THRESHOLD.



EXPANSION RATIO
Range: 1/1 to 1/40, INF
This parameter sets the amount of expansion. The range is based on decibels (dB) below the threshold. If this is set to 1/4 for example, it will expand changes in signals below the threshold by a factor of four. When this is set to infinity, it acts as a gate. A setting of 1/1 offers no expansion.

THRESHOLD
Range: -96 to +0 dB
This parameter sets the threshold level. Signals that exceed this level will be unaffected, while signals that are below will be expanded. To turn off the expander, set the level to -96 dB.

ATTACK TIME
Range: 50µs to 100ms
This parameter determines the attack rate after the initial signal has been detected and before the expansion takes affect.

RELEASE TIME
Range: 1ms to 10.0s
This parameter determines the release rate after the signal has been detected below the threshold level. This is generally chosen longer than the ATTACK TIME.

HOLD TIME
Range: 1ms to 10.0s
This parameter is the detection sustain time in the ADSR that constitutes attack, sustain, and release.

EQ GAIN
Range: -48 to +48 dB
This parameter controls the amount of boost applied to the output signal of the high/low pass filter. This accounts for insertion loss through those filters.

HIGH PASS FC
Range: 4 to 8000 Hz
This sets the cutoff frequency of the lower frequency band high pass shelving filter.

LOW PASS FC
Range: 100 Hz to 15 KHz
This parameter sets the amount of boost or cut applied to the low pass filter.

TRIGGER MASK
Range: OFF or ON
This parameter enables the trigger mask function. Once triggered, the side chain detector will see no input signal for a duration specified by the TRIGGER TIME.

TRIGGER TIME
Range: 1ms to 10.0s
This parameter sets the duration over which the side chain detector will be blacked out. This parameter is useful for isolating the first bar of a drum track.

MASK THRESHOLD
Range: -96 to +0 dB
This parameter sets the trigger mask threshold level. Signals that fall below this level will trigger the mask function. The trigger mask function uses the (expander) THRESHOLD as upward hysteresis. Therefore, the MASK THRESHOLD should always be set lower than the (expander) THRESHOLD.

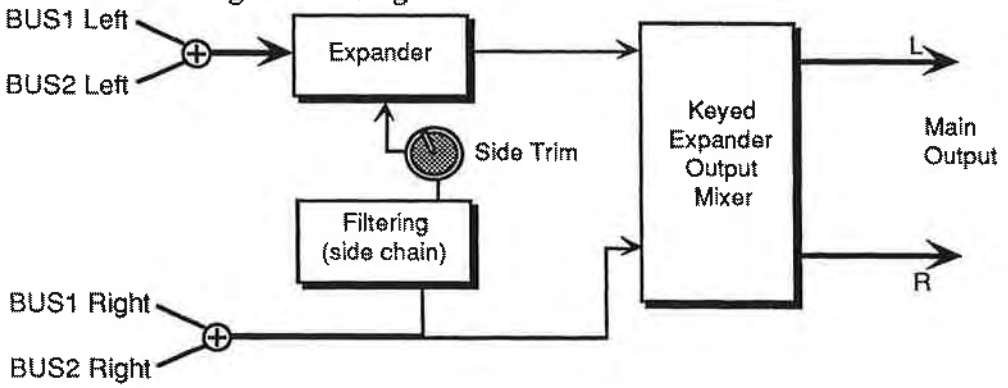
OUTPUT GAIN
Range: -48 to +48 dB
This parameter sets the amount of cut (negative values) or boost (positive values) applied to the expander on the output volume. We recommend a starting application of +0 dB.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-44 KEYED EXPNDR

KEYED EXPNDR operation is identical to the Expander. The only difference is that the left signal (BUS1 or BUS2 hard left) is expanded as determined by the key. The key is the right channel signal (BUS1 or BUS2 hard right). Make sure your sources are routed this way in order for this effect to work properly. BUS3 is a dry path.

KEYED EXPANDER Signal Routing



EXPANSION RATIO
Range: 1/1 to 1/40, INF
This parameter sets the amount of expansion. The range is based on decibels (dB) below the threshold. If this is set to 1/4 for example, it will expand changes in signals below the threshold by a factor of four. When this is set to infinity, it acts as a gate. A setting of 1/1 offers no expansion.

THRESHOLD
Range: -96 to +0 dB
This parameter sets the threshold level. Signals that exceed this level will be unaffected, while signals that are below will be expanded. To turn off the expander, set the level to -96 dB.

ATTACK TIME
Range: 50µs to 100ms
This parameter determines the attack rate after the initial signal has been detected and before the expansion takes affect.

RELEASE TIME
Range: 1ms to 10.0s
This parameter determines the release rate after the signal has been detected below the threshold level. This is generally chosen longer than the attack time.

HOLD TIME
Range: 1ms to 10.0s
This parameter is the detection sustain time in the ADSR which constitutes attack, sustain, and release.

EQ GAIN
Range: -48 to +48 dB
This parameter controls the amount of boost applied to the output signal of the high/low pass filter. This accounts for insertion loss through those filters.

HIGH PASS FC
Range: 4 to 8000 Hz
This parameter sets the cutoff frequency of the lower frequency band high pass shelving filter.

LOW PASS FC
Range: 100 Hz to 15 KHz
This parameter sets the amount of boost or cut applied to the low pass filter.

TRIGGER MASK
Range: OFF or ON
This parameter enables the trigger mask function. Once triggered, the side chain detector will see no input signal for a duration specified by the TRIGGER TIME.

TRIGGER TIME
Range: 1ms to 10.0s
This parameter sets the duration over which the side chain detector will be blacked out. This parameter is useful for isolating the first bar of a drum track.

MASK THRESHOLD
Range: -96 to +0 dB
This parameter sets the trigger mask threshold level. Signals that fall below this level will trigger the mask function. The trigger mask function uses the (expander) THRESHOLD as upward hysteresis. Therefore, the MASK THRESHOLD should always be set lower than the (expander) THRESHOLD.

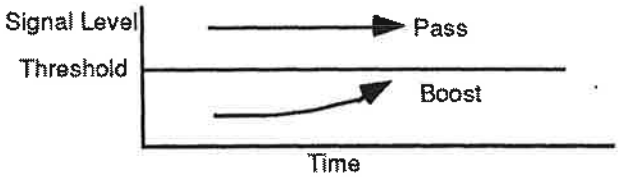
OUTPUT MIX
Range: 0 to 99
This parameter mixes the output of the left signal with the output of the right signal. This is the output mixer that is shown in the diagram.

OUTPUT GAIN
Range: -48 to +48 dB
This parameter sets the amount of cut (negative values) or boost (positive values) applied to the expander on the output volume. We recommend a starting application of +0 dB.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-45 INVRS EXPNDR

INVRS EXPNDR creates sustain by expanding the signal so that the signal levels above threshold are passed and levels below threshold are boosted to create a more even sound. A traditional expander would have the opposite effect: that is a signal level below threshold would be attenuated. An inverse expander is much like a compressor in so far as they both can be used to create sustained sounds, and de-emphasize transient signals. EQ exists in both signal and side chain paths, in contrast to the Expander that has filtering in only the side chain path.



The parameters available for this algorithm are:

EXPANSION RATIO
Range: 1/1 to 40/1, INF
This parameter sets the amount of expansion. The range is based on decibels (dB) below the threshold. If this is set to 3/1 for example, it will expand the changes in signals that are below the threshold level by three times. We recommend starting with settings near 1/1 (a setting of exactly 1/1 disables expansion).

THRESHOLD
Range: -96 to +0 dB
This parameter sets the threshold level. Signals below this level will be boosted, while signals that are above will be unaffected. As the input signal dies away below the threshold, the expander will increase the signal gain. To turn off the inverse expander, set the threshold to -96dB.

ATTACK TIME
Range: 50us to 100ms
This parameter determines the time after the initial signal amplitude has been detected for the expansion to take affect.

RELEASE TIME
Range: 1ms to 10.0s
This parameter determines how long it takes for the expansion to be fully deactivated after the input signal rises above the threshold level. This is generally longer than the ATTACK TIME.

GATE OFF
Range: -96 to +0 dB
This parameter sets the lower threshold level at which the noise gate shuts off the audio.

GATE ON
Range: -96 to +0 dB
This parameter sets the upper threshold level at which the noise gate passes audio. This second parameter provides hysteresis.

BASS FC
Range: 0 to 1000 Hz
This parameter sets the cutoff frequency of the lower frequency band shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the low shelving filter.

TREBLE FC
Range: 1KHz to 15KHz
This parameter sets the cutoff frequency of the upper frequency band high shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the high shelving filter.

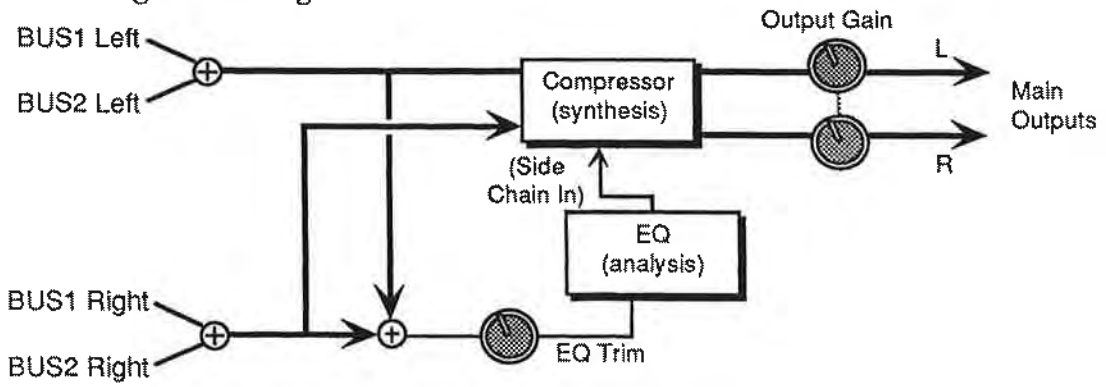
EQ TRIM
Range: -24 to +0 dB
This parameter allows you to adjust the level of the signal entering the EQ, to eliminate the possibility of clipping boosted signals.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-46 DE-ESSER

DE-ESSER is a stereo algorithm that compresses sibilant frequencies (like the "ess" sound) as they become louder. This algorithm was designed for vocalists, but it can also be used to control the boomy sound of a guitar or the ringing sound of drums by adjusting the side-chain appropriately.

DE-ESSER Signal Routing



The parameters available for this algorithm are:

OUTPUT GAIN
Range: -48 to +48 dB
This parameter sets the amount of cut (negative values) or boost (positive values) applied to the de-esser on the output volume. We recommend a starting application of +0 dB.

COMP RATIO
Range: 1/1 to 40/1, INF
This parameter sets the amount ot compression. The range is based on decibels (dB) above the threshold. If this is set to 4/1 for example, it will compress changes in signals above the threshold by one quarter. When this is set to infinity, it acts as a limiter.

THRESHOLD
Range: -96 to +0 dB
This parameter sets the threshold level. Signals that exceed this level will be compressed, while signals that are below will be unaffected.

ATTACK TIME
Range: 50µs to 100ms
This parameter determines the attack rate after the initial signal has been detected and before the compression takes affect.

RELEASE TIME
Range: 1ms to 10.0s
This parameter determines how long it takes for the compression to be fully deactivated after the input signal drops below the threshold level. This is generally chosen longer than the attack time.

GATE OFF
Range: -96 to +0 dB
This parameter sets the lower threshold level at which the noise gate shuts off the audio.

GATE ON
Range: -96 to +0 dB
This parameter sets the upper threshold level at which the noise gate passes audio. This higher second threshold prevents false "turn ons."

HIGH PASS FC
Range: 4 to 8000 Hz
This parameter controls a hi pass filter frequency for the side chain EQ. This is useful for de-essing.

BASS FC
Range: 0 to 1000 Hz
This parameter sets the cutoff frequency of the lower frequency band shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the low shelving filter.

MID1 FC
Range: 0 to 9999 Hz
This parameter sets the center of the mid-frequency parametric.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to this frequency parametric.

MID1 Q
Range: 1 to 18
This parameter is a bandwidth control that determines the width of the resonant peak at the center of the frequency band. By raising the band you can produce a narrower bandwidth.

MID2 FC
GAIN
MID2 Q
These three parameters are identical to the previous three parameters, and can be used to control different bandwidths within the mid range.

TREBLE FC
Range: 1KHz to 15KHz
This parameter sets the cutoff frequency of the upper frequency band high shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the high shelving filter.

EQ TRIM
Range: -48 to +0 dB
This parameter adjusts the input level to the side chain EQ, which performs analysis on the input signal to selectively compress it.

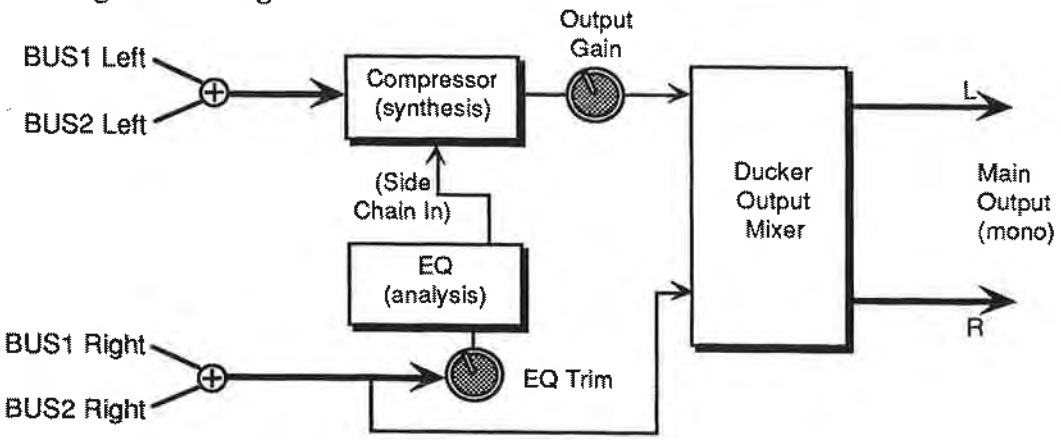
MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-47 DUCKER

The DUCKER algorithm is a compressor that automatically lowers the level of a signal (such as music) when another signal (like an announcer voice-over) comes in. When the voice-over leaves, the level of the original signal is restored. This algorithm is useful for voice-overs, Rap, and DJ work. In order for this algorithm to work properly, the music (audio source to be ducked) must be sent to BUS1 or BUS2 hard left, and the voice-over is sent to BUS1 or BUS2 hard right. In this set up, BUS1 or BUS2 (hard right) is considered the side chain to a traditional compressor. This algorithm employs an internal mixer that mixes left and right inputs to a mono output.

The gate function is achieved for high compression ratios. In this application a transient signal source, such as a snare drum, can gate on and off some other music signal, to achieve an externally controllable staccato effect.

DUCKER Signal Routing



OUTPUT GAIN
Range: -48 to +48 dB
This parameter sets the amount of cut (negative values) or boost (positive values) applied to the ducker on the output volume. We recommend a starting application of +0 dB.

OUTPUT MIX
Range: 0 to 99
This parameter mixes the output of the music signal (BUS1 or BUS2 hard left) with the output of the ducker (voice-over) signal (BUS1 or BUS2 hard right) to a mono output. This is the Ducker Output Mixer that is shown in the diagram.

COMP RATIO
Range: 1/1 to 40/1, INF
This parameter sets the amount of compression. The range is based on decibels (dB) above the threshold. If this is set to 4/1 for example, it will compress changes in signals above the threshold by one quarter. When this is set to infinity, it acts as a limiter.

THRESHOLD
Range: -96 to +0 dB
This parameter sets the threshold level. Signals that exceed this level will be compressed, while signals that are below will be unaffected.

ATTACK TIME
Range: 50µs to 100ms
This parameter determines the attack rate after the initial signal has been detected and before the compression takes affect.

RELEASE TIME
Range: 1ms to 10.0s
This parameter determines how long it takes for the compression to be fully deactivated after the input signal drops below the threshold level. This is generally chosen longer than the ATTACK TIME.

GATE OFF
Range: -96 to +0 dB
This parameter sets the lower threshold level at which the noise gate shuts off the audio.

GATE ON
Range: -96 to +0 dB
This parameter sets the upper threshold level at which the noise gate passes audio. This higher second threshold prevents false "turn ons."

BASS FC
Range: 0 to 1000Hz
This parameter sets the cutoff frequency of the lower frequency band shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the low shelving filter.

MID1 FC
Range: 0 to 9999 Hz
This parameter sets the center of the mid-frequency parametric.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to this frequency parametric.

MID1 Q
Range: 1 to 18
This parameter is a bandwidth control that determines the width of the resonant peak at the center of the frequency band. By raising the band you can produce a narrower bandwidth.

MID2 FC
GAIN
MID2 Q
These three parameters are identical to the previous three parameters, and can be used to control different bandwidths within the mid range.

TREBLE FC
Range: 1KHz to 15KHz
This parameter sets the cutoff frequency of the upper frequency band high shelving filter.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to the high shelving filter.

HIGH PASS FC
Range: 4 to 8000 Hz
This parameter controls the boost or cut of the high pass filter frequency applied to the input signal. This is useful for ducking.

EQ TRIM
Range: -48 to +0 dB
This parameter adjusts the input level to the side chain EQ, which performs analysis on the input signal to selectively compress it.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-48 RUMBLE FILTR

RUMBLE FILTR is a high pass filter in cascade with a low pass filter, fourth order (24dB per octave). The high pass filter is good for eliminating turntable rumble. The low pass filter is good for eliminating hiss. Alternatively, these filters can be used in a feedback routing with any other effect. The BUS1, BUS2, and BUS3 MIX parameters can be routed into the effect with different Dry/Wet mixes. For this algorithm, we recommend mid values of the Mix.

The parameters available for the Rumble Filter are:

HIGH PASS FC
Ranges: 4 to 8000 Hz
This parameter controls the boost or cut of the high pass filter frequency applied to the input signal.

LOW PASS FC
Ranges: 100 Hz to 15 KHz
This parameter controls the boost or cut of the low pass filter frequency applied to the input signal.

FILTER GAIN
Ranges: -48 to +48 dB
Because the cascade of high pass with low pass causes an insertion loss, this parameter allows you to boost the filtered output signal.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-49 PARAM EQ

PARAM EQ offers a minimum phase four band parametric EQ. The parameters available for this algorithm are:

BAS FC
Range: 0 to 1000 Hz
This parameter sets the center of the low frequency parametric.

GAIN
Range: -48 to +24 dB
This sets the amount of boost or cut applied to this low frequency parametric.

MID1 FC
Range: 0 to 9999 Hz
This parameter sets the center of the mid frequency parametric.

GAIN
Range: -48 to +24 dB
This sets the amount of boost or cut applied to this mid frequency parametric.

MID1 Q
Range: 1 to 18
This parameter is a bandwidth control that determines the width of the resonant peak at the center frequency band. By raising the value you can produce a narrower bandwidth.

MID2 FC
GAIN
MID2 Q
These three parameters are identical to the previous three parameters, and are used to control different bandwidths within the mid range.

TREBLE FC
Range: 1 to 15 KHz
This parameter sets the center frequency of the high frequency parametric.

GAIN
Range: -48 to +24 dB
This parameter sets the amount of boost or cut applied to this high frequency parametric.

EQ TRIM
Range: -24 to +0 dB
This parameter allows you to adjust the input level trim to the EQs to eliminate the possibility of clipping boosted signals.

MOD SRC 1
DST 1
MIN 1 / MAX 1
MOD SRC 2
DST 2
MIN 1 / MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

ROM-50 VAN DER POL

VAN DER POL Filter adds synthetic high harmonics to the input signal, brightening the overall sound. This newly designed algorithm is most often used in the studio for vocalists, but feel free to experiment with this algorithm using your favorite instrument as well. This algorithm features prominent transient enhancement that makes it ideal for "plucked" sounds. The filter in this algorithm operates on the signal prior to enhancement. Set the filter to enhance the frequency region that you desire. Then use the BUS1 MIX parameter to mix the enhanced signal with the dry signal. For this algorithm we recommend mid values of the Mix.

The parameters available for the Van Der Pol Filter are:

HIGH PASS FC
Ranges: 4 to 8000 Hz
This parameter controls the boost or cut of the high pass filter frequency applied to the input signal.

LOW PASS FC
Ranges: 100 Hz to 15 KHz
This parameter controls the boost or cut of the low pass filter frequency applied to the input signal.

FILTER GAIN
Ranges: -48 to +48 dB
Because the cascade of high pass with low pass causes an insertion loss, this parameter allows you to boost the filtered output signal.

MOD SRC 1
DST 1
MIN 1
MAX 1
MOD SRC 2
DST 2
MIN 2
MAX 2
These parameters are explained in detail in *Section 5 — Effect Concepts*.

Sampling Signal Source Concepts

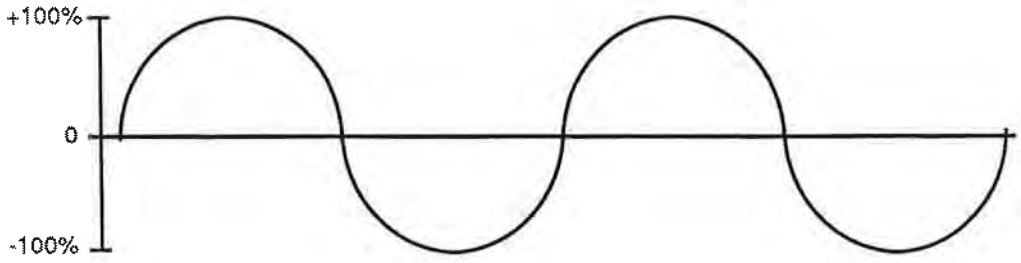
Section 7 — Sampling/Signal Source Concepts

This section provides an overview of the sampling process. For a complete list of the sampling parameters, refer to the following section.

What is Sampling?

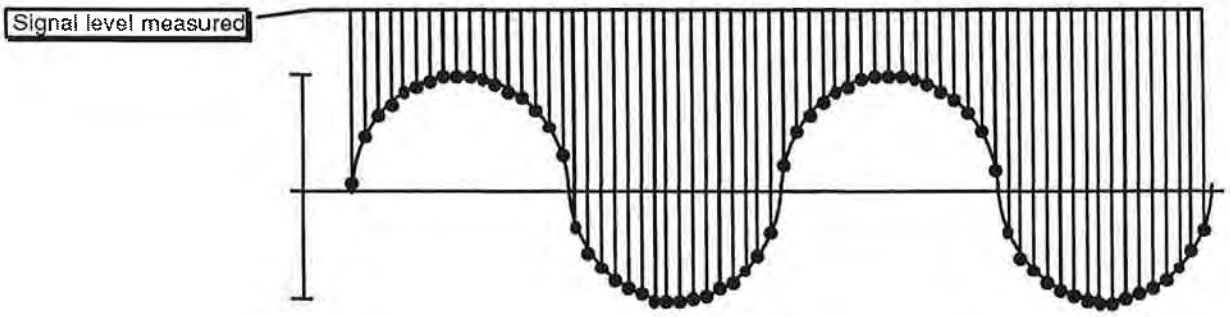
Before getting into talking about samples and sampling, let's begin with a simple explanation of sound. A sound begins as a series of vibrations, or pressure waves, in the air. When these vibrations reach the diaphragm of a microphone, they cause it to move back and forth. This creates a fluctuating electrical signal that rises and falls around a center, or zero line.

A simple sound wave, once it has been converted into an electrical signal, might look like this:



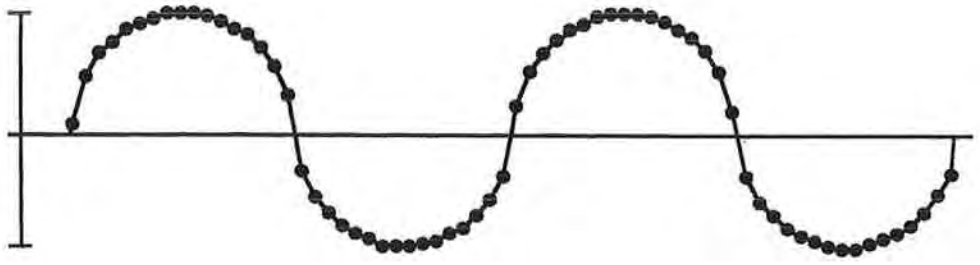
A conventional analog tape recorder would record this signal by converting the electrical fluctuations into magnetic fluctuations and then imprinting a continuous (or "analog") copy onto the magnetic surface of the tape.

A digital sampling system, such as the ASR–10 or a digital tape recorder, works a little differently. When you digitally record the sound, the level of the signal is measured (or "sampled") thousands of times per second, and each number is recorded as a number in memory. If the same signal that was shown in the analog recording illustration above was recorded digitally, it would look like this:



The digital sampler does not record the actual sound but rather encodes a series of discrete numbers, each of which represents the level of the signal at a given instant in time.

On playback, the ASR–10 reconstructs the original signal by "connecting the dots," producing an output voltage that corresponds to the numbers in its memory (shown below). This voltage can then be amplified and sent to speakers, which turn it back into pressure waves in the air so that we can hear it.



Sample Rate

The sample rate determines how often the signal is measured (that is, how close together the dots would be in the illustration above). A sample rate of 30 kHz, for example, means that the ASR–10 is sampling the signal 30,000 times each second. A higher sample rate will yield a more accurate picture of the wave form and better high frequency response. But keep in mind that a higher sample rate also will use up the ASR–10 memory more quickly, since you are generating more numbers per second that have to be stored in memory.

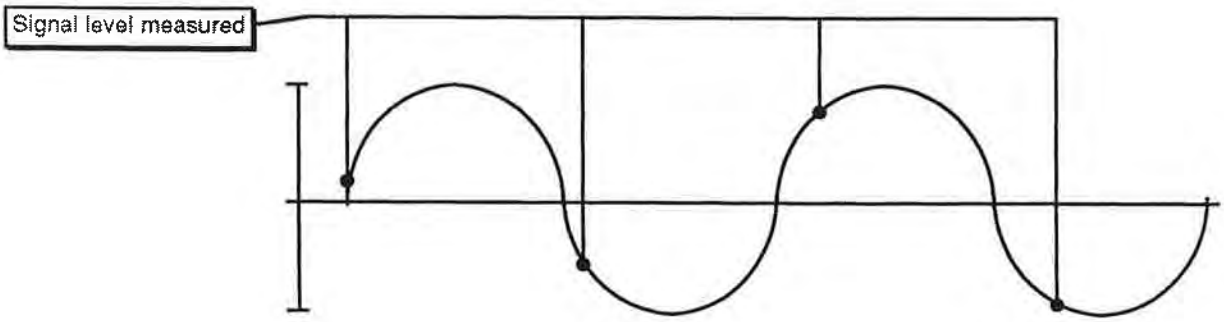
The ASR–10 can sample at either 30 kHz (actually 29,761.9 samples per second) or 44 kHz (44,100.0 samples per second). To give you some frame of reference, Compact Discs are also recorded at 44.1 kHz, and Digital Audio Tape (DAT) is recorded at 48 kHz. You can adjust the ASR–10 sample rate by changing the current effect algorithm on the FX Select•FX Bypass page.

Note: When the current system sample rate is 44.1 kHz, the Digital Output will provide direct digital audio output of the Main Output mix, when the optional DI-10 Digital I/O Board is installed.

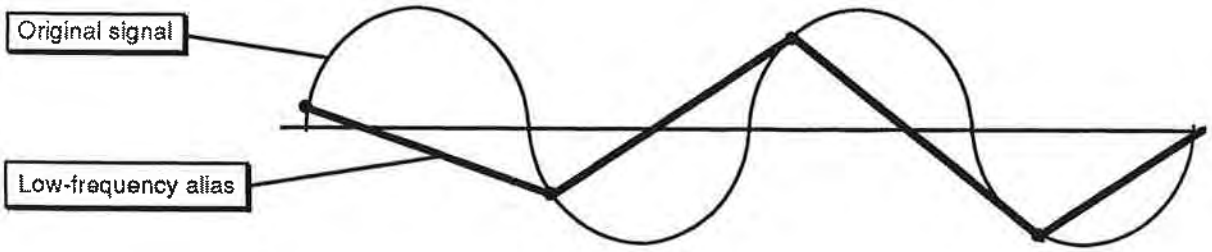
Aliasing

A digital sampler can only accurately record frequencies up to one half that of the sample rate. This frequency (half the sample rate) is known as the *Nyquist frequency*, named after the fellow who figured that out. Trying to sample sounds that contain frequencies above half the sample rate will result in *aliasing*, the creation of false (or "alias") low-frequency tones, when playing the sound back.

Suppose we were to sample the wave from our previous example at a much lower sample rate (fewer times per second), like this:



When the ASR–10 tries to reproduce this sound (to connect the dots, as it were) the resulting wave bears little resemblance to the original, because there are not enough sample points to accurately recreate it:



To prevent aliasing, the ASR–10 has a 64 times oversampling, one-bit Sigma-Delta, stereo A/D input converter, which offers superior audio fidelity.

Getting Ready to Sample

Here are a few things you can do that will result in your getting better quality samples.

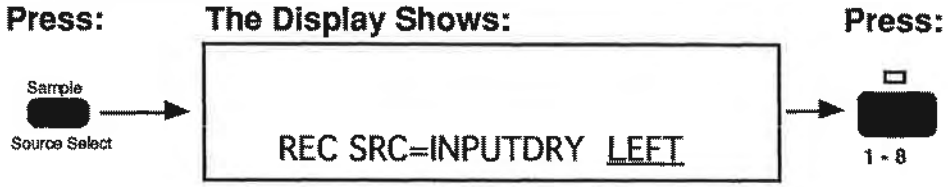
- **Start with the cleanest possible signal** — This may seem obvious, but a sample (or any recording) can only be as good as the original signal, so you should pay attention to the signal path. For example, if you are feeding the audio input of the ASR–10 from a mixer, using a direct out from the individual channel is better than using an auxiliary send because there are fewer preamps (and thus less noise) in the chain. If you do use an auxiliary or a main send, make sure *only* the channel(s) you intend to sample are turned up in that send.
- **Record the sound on tape and sample from the tape** — This is not always possible, but it tends to yield better results because now the sound is repeatable. This is important when adjusting the input level — before sampling you can play the sound as many times as necessary, watching the meter and adjusting the level of the signal into the ASR–10, until you have just the right level — high enough to get the best possible dynamic range, yet low enough to avoid clipping. The other good reason for sampling from tape also has to do with repeatability. Suppose you are sampling a horn section live, directly into the ASR–10, and suddenly they play that one excruciatingly perfect note that you've been waiting for all your life. But after making the sample you discover that the input level was a little too high and the sample is clipped. Or, you want to put a slightly different EQ on it, or maybe some reverb. Unfortunately, you will never get them to play that note just the same way again, so the moment is lost. If you had it on tape, you could simply make your adjustments and sample it again.

Easy Sampling

Let's try a simple one-shot sample. First we need a source. For example, a guitar — but any source, such as a synthesizer, the output of a mixer or cassette player, etc. could be used. That sound source should be connected to one of the Audio Input jacks of the ASR–10.

Adjust the ASR–10 sample rate by changing the current effect algorithm on the FX Select•FX Bypass page (or select 23 VOICE 44K or 31 VOICE 30K to monitor the signal source dry). A higher sample rate will yield a more accurate sample with better high frequency response, but will use up memory more quickly. The ASR–10 can sample at either 30 kHz (actually 29,761.9 samples per second) or 44 kHz (44,100.0 samples per second).

To initiate sampling press the ***Sample•Source Select*** button to the right of the display. The display shows:



The RECORD SOURCE parameter has 2 fields. Field 1 determines which Audio Source will be sampled. Field 2 determines whether the Audio Source will be sampled in stereo or mono.

The available REC SRC Field 1 values are:

- INPUTDRY — This is the base setting, and it allows you to sample Audio Inputs dry (with no effect). ***Audio Track A*** monitors the left channel of the REC SRC, and ***Audio Track B*** monitors the right channel. You can hear how the dry signal will sound through the currently selected effect as it is monitored on the Edit/(audio) Track OUT bus.
- INPUT+FX — This value allows you to sample Audio Inputs wet (with the currently selected effect) on BUS1. All other voices are dry. Any WaveSamples, Instrument•Sequence Tracks or Audio Tracks that were routed to BUS1, 2, or 3 will all show OUT=DRY- -FX BUS IN USE on their Edit/Amp or Edit/Track, OUT bus screens.

  When REC SRC= INPUT+FX, the ESP chip is dedicated to processing the Audio Inputs, and the audio tracks monitor the direct output of the ESP chip. The left effects channel is monitored on ***Audio Track A*** and the right channel on ***Audio Track B***. The Wet/Dry mix is set on the Edit/Effects page. Note that when the current effect is gain-dependent (for example distortion or compression), the effect output may not sound the same as it would if REC SRC= INPUTDRY. This happens because when the REC SRC= INPUT+FX, the audio input source is routed directly into the effects processor at full level. To compensate, adjust the volume of the external audio source.

- MAIN-OUT — This allows you to resample any ASR–10-generated audio routed to OUT BUS1, 2, or 3. In VU mode, instruments can be Selected and Stacked in any combination before sampling. The "pile" is playable from the controller keyboard via MIDI. You can also press ***Play*** to start the sequencer and sample sequences and songs. The current effect also remains unchanged. For the optimal signal-to-noise ratio, the Volume Slider should be at maximum when sampling the MAIN-OUT.

Hint: When REC SRC= MAIN-OUT, you can use the volume slider in real time to sample volume fade-ins and outs.

Hint: After sampling the MAIN OUT, the Edit/Amp BOOST parameter is automatically set to ON in order to match the level between the original WaveSample and the resampled WaveSample. As a final step, after you're happy with the resampled results, you can futher optimize the WaveSample by setting the Edit/Amp BOOST parameter to OFF, and performing the Command/Amp NORMALIZE GAIN command.

- DIGITAL — Sample the Digital Input (requires the optional DI-10 Digital I/O interface). If no digital interface is installed, this setting is not available.

The available REC SRC Field 2 values are:

- LEFT — Sample the LEFT channel of the REC SRC in MONO
- RIGHT — Sample the RIGHT channel of the REC SRC in MONO
- L+R — Sample both the LEFT and RIGHT channels of the REC SRC simultaneously in STEREO

Using the Audio Tracks During Sampling

Pressing the ***Audio Track*** buttons will also change the REC SRC Field 2 value to reflect the current combination of selected Audio Tracks. For instance, if you press the ***Audio Track A*** button, the REC SRC Field 2 value changes to LEFT. If you press ***Audio Track B***, the REC SRC Field 2 value changes to RIGHT. If you press ***Audio Track A*** and while holding it down press ***Audio Track B***, the REC SRC Field 2 value changes to L+R. When both Audio Tracks are selected (both yellow LEDs lit), holding one ***Audio Track*** button and pressing the other one will change the REC SRC Field 2 value to LEFT or RIGHT, deselecting *both* Audio Tracks. The REC SRC Field 2 value will be set to reflect the ***Audio Track*** button that was pressed first. Both Audio Tracks will become Source Monitor disabled (no red LEDs lit).

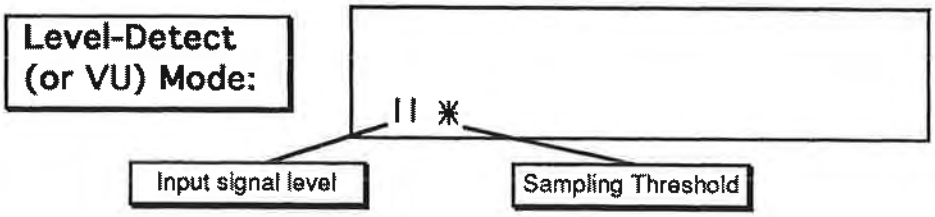
For more information about Selecting, Enabling, and Disabling Audio Tracks, see *Section 12 — Sequencer and Audio Track Concepts*.

Note: Whenever the ***Sample•Source Select*** button is pressed, the current FX settings remain unchanged.

Once you've selected the Record Source field values for your signal source, press either an ***Instrument•Sequence Track*** button (the recommended way to select a destination Instrument) *or* press the ***Enter•Yes*** button. If the ***Enter•Yes*** button is pressed, the display will ask:

- Press an unused (LED not lit) ***Instrument•Sequence Track*** button. If you select an ***Instrument•Sequence Track*** where there is already an instrument loaded, the ASR–10 goes to the multisampling screen. Don't do that yet.

After you've pressed an unused ***Instrument•Sequence Track*** button, the ASR–10 goes into Level-Detect (or VU) mode:



In Level-Detect mode the display acts like a peak-reading VU meter, the vertical bars on the display lighting from left to right show the level of the incoming signal. The *star* represents the sampling threshold — the ASR–10 will not begin recording until the signal crosses this level. The Sampling Threshold provides a way to make the ASR–10 "wait" until the signal reaches a certain level before it begins to record. The ASR–10 offers twenty possible sample threshold levels. Pressing the ***Up Arrow*** button raises the threshold; pressing the ***Down Arrow*** button lowers it.

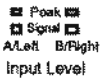
Hint: While in Level-Detect Mode, you can select, deselect and stack (double-click) ***Instrument•Sequence Tracks***, and can play them from an external MIDI controller, even during sampling. You can also press ***Play*** and ***Stop•Continue*** to sample along with sequence playback, or even to sample sequence playback when REC SRC= MAIN-OUT. This way, what you are about to sample can "sync-up" to what you've already sequenced or sampled (pitch, timing, etc.). If you scroll right or left to view any of the other sampling screens, you will only hear the destination instrument (the instrument into which you are sampling).

Before you proceed, there are a number of sampling parameters you can adjust. These parameters are explained in detail in *Section 8 — Sampling/Signal Source Parameters*.

Setting the Input Level

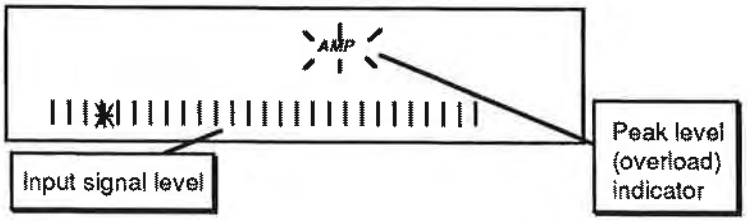
As mentioned previously, optimizing the input level is crucial in sampling. Record at too low a level and you don't use all the bits of resolution that are available in the system — the sample will be too quiet and overly noisy. Record at too high a level and you will introduce clipping, which tends to be extremely harsh and unpleasant. For the sake of this and the examples that follow, we will assume that you are using an input signal from a repeatable source, such as a tape player, record, CD or synthesizer, whose level you can adjust.

- When REC SRC= INPUTDRY or INPUT+FX, the software VU meter is supplemented by two hardware Signal and Peak Input Level LED meters. These two dedicated Input Level meters provide separate Left/Right metering of the Audio Input levels, pre-FX, at all times. The green Signal LEDs light at -24 dB. The red Peak LEDs light at 6 dB below clipping.



- Play the signal into the ASR–10. Watch the display to see if the AMP indicator lights (this light indicates overload, or clipping). If it does light, use the ***Input Level Trim Control*** found on the rear panel to adjust the signal. Now play the sound into the ASR–10 again, watching the indicator and adjusting the signal accordingly.

- Repeat this until you have found the point just barely *below* where the AMP indicator lights. This is the optimum recording level. With a little practice you can quickly get a good level every time.



Once you have the level adjusted:

- Press ***Enter•Yes*** or press the Left Foot Switch to initiate sampling. The display will show WAITING... until the threshold is exceeded. The AMP annunciator will light if the Input Source clips. During sampling the display shows:

The xxx value (range 000 to 999) will count down in real time when sampling. Press any button to stop sampling. Remember that if the sampling threshold (the star) is set higher than the input signal level (the vertical bars), the ASR–10 will not begin sampling until the input signal crosses the threshold. Pressing ***Cancel•No*** while the sampling screen reads WAITING will abort sampling, and the ASR–10 will return to Level–Detect mode.

- Play the sound to be sampled; play the guitar, synth, start the recording, or speak into the microphone. If you are planning on using a microphone, we recommend plugging it into a mixer or preamp, and sending a line from the mixer/preamp to the Left Audio Input jack, for optimal performance.
- Press ***Cancel•No*** or press the Left Foot Switch to stop sampling. If you don't stop it by pressing ***Cancel•No*** or the Left Foot Switch, the ASR–10 will continue sampling until it has used up all the available memory.

Note: In order to use the foot switch to start and stop sampling, the LEFT FOOT SW parameter on the Edit/System•MIDI page must be set to LEFT FOOT SW= SAMPL YES.

After you have stopped sampling (or the memory is full) the display flashes PLAY ROOT KEY OR ENTER. The Root Key is the incoming MIDI note number that will play the WaveSample at its original pitch. In other words, if you play a C4 on the MIDI keyboard connected to the ASR–10, the sample will play back at "unity" (the same pitch as the original input signal) whenever the C4 key is pressed. Whichever key you press in response to the PLAY ROOT KEY OR ENTER prompt is the key that will play back exactly what you sampled.

Hint: You can also press the ***Enter•Yes*** button in response to the PLAY ROOT KEY OR ENTER prompt. This will set the root key of the new WaveSample to the same note assigned to the ENTER PLAYS Key parameter, and allows you to record samples using your ASR–10 without being connected to a MIDI keyboard. Remember, you can assign different WaveSample ranges and change the pitch using the Edit/Instrument parameters.

Normalizing Gain

After you've recorded a satisfactory sample, you should then perform the Command/Amp NORMALIZE GAIN command, to optimize use of the full dynamic range of the ASR–10. To do this:

- Press ***Command***, then ***Amp*** and scroll until the display reads NORMALIZE GAIN.
- Press ***Enter•Yes*** to invoke the command. The display will read DATA BEING PROCESSED, followed by COMMAND COMPLETED.

For more information on the NORMALIZE GAIN command, see *Section 10 — Wave Data Parameters*.

Default Output Routings

- When REC SRC= MAIN-OUT, and a new WaveSample is created, its Edit/Amp, OUT value will be set to BUS3 (Dry).
- When REC SRC= INPUT+FX, and a new WaveSample is created, its Edit/Amp, OUT value will be set to BUS3 (Dry).
- When REC SRC= INPUTDRY, and a new WaveSample is created, its Edit/Amp, OUT value will be set to the same value as the Edit/(audio) Track, OUT parameter for the Audio Track through which the RECORD SOURCE was monitored.
- When REC SRC= DIGITAL, and a new WaveSample is created, its Edit/Amp, OUT value will be set to the same value as the Edit (audio) Track, OUT parameter for the Audio Track through which the RECORD SOURCE was monitored.

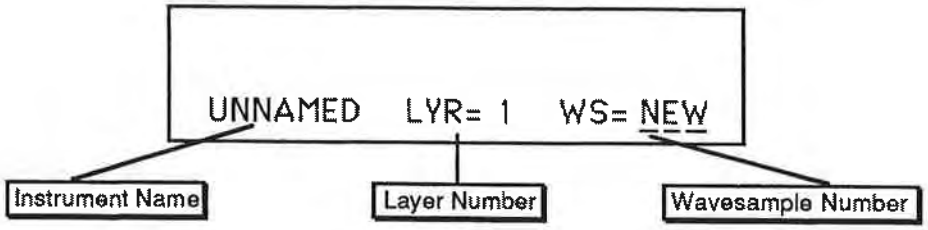
MultiSampling

Many sounds require that you make more than one sample, with each sample covering a different range of the instrument. This is because if you transpose a sound too far (up or down) from its original pitch it begins to sound unnatural. Another application for making multiple samples is when you are sampling something like a drum set, where you want several entirely different sounds on the keyboard at once.

The ASR–10 makes it easy to "multisample" — which is what we call this process of making several samples within one Instrument.

- Make your first sample as described above in "Easy Sampling."
- Press the ***Sample•Source Select*** button.
- Press the ***Instrument•Sequence Track*** button of the instrument you just sampled.

When you select an existing instrument from the REC SRC screen, the following multisampling screen (also called the Sampling Context screen) appears:



Here you see:

- The first eight letters of the instrument name (which is always UNNAMED for newly sampled sounds);
- The number of the current layer; and
- Which WaveSample will be sampled after you press ***Enter•Yes***. The default here is NEW, meaning that a new WaveSample will be created in the same layer of the instrument.
- Press ***Enter•Yes***. This puts you back in Level-Detect mode. Now proceed to sample the new WaveSample just as you did the first one:
- Adjust the incoming level so that it doesn't clip.
- Press ***Enter•Yes*** or the Left Foot Switch to begin sampling.
- Play the sound to be sampled.
- Press ***Cancel•No*** to stop sampling.
- Play the new Root Key — the key at which you want the new sample to play back.

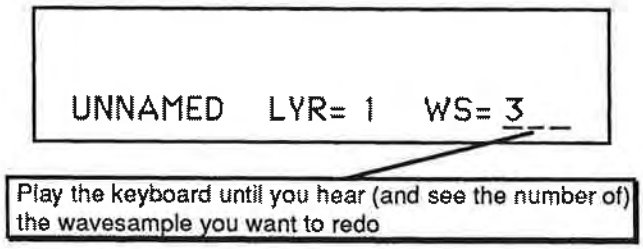
Once you have assigned a Root Key to the new sample, the ASR–10 automatically adjusts the split point to be midway between the Root Keys of the two samples.

Follow the exact same procedure to continue making multiple samples within the instrument.

Sampling Over an Existing WaveSample

Let's suppose you have made four multisamples of some instrument, and you want to go back and redo the third one. The procedure is almost the same as that for making a new multisample: instead of pressing ***Enter•Yes*** while WS=NEW, you will first pick one of the existing WaveSamples by selecting it from the MIDI keyboard connected to the ASR–10.

- Press ***Sample•Source Select***, then select the instrument into which you want to sample.
  The multisampling screen appears as before.
- Move the cursor (underline) to WS =__
  The cursor should be beneath the WaveSample Select parameter (WS=__). If it's not, use the ***Right Arrow*** button move it there.
- Play a note on the MIDI keyboard connected to the ASR–10 to select the WaveSample to redo. As you play notes you will see the word NEW replaced by the number of the WaveSample you are hearing. As you play different notes, the WaveSample number is updated. When you hear the one you want to redo, and see its number on the display, it is selected to be sampled over. Or you can scroll to the WaveSample number you want to redo using the ***Up/Down Arrow*** buttons, or the ***Data Entry Slider***.



- Press ***Enter•Yes*** and continue with sampling as before. The original WaveSample will be erased, and will be replaced by the new one.

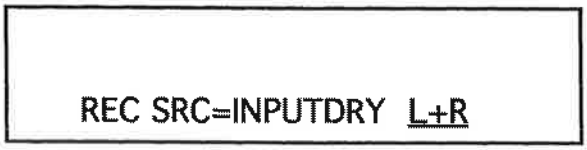
When you sample over an existing WaveSample, the ASR–10 will preserve the keyboard range, all WaveSample parameters (envelopes, etc.) and, when possible, the loop points of the original sample.

Note: When you sample into a layer that already contains WaveSamples, the existing WaveSamples in that layer will not be audible if the Instrument is selected in VU mode.

Stereo Sampling

The ASR–10 offers true stereo sampling, through Effects if desired. The ASR–10 also allows real time stereo resampling of notes played on the keyboard, or received via MIDI. Here's how to sample in stereo:

- Press the ***Sample•Source Select*** button. The display shows:

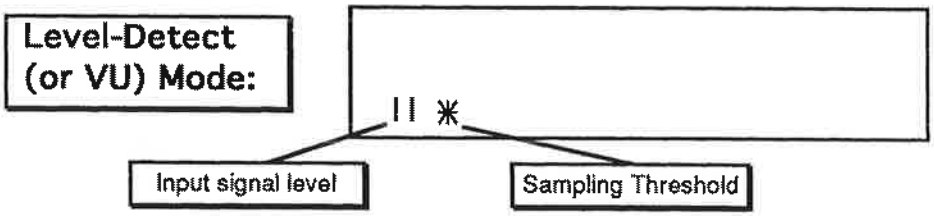


When only one Audio Track is Selected, holding one Audio Track button and pressing the other one will change the REC SRC Field 2 value to L+R (as shown above), selecting both Audio Tracks and enabling stereo sampling. The Audio Track button that is held (i.e. pressed first) will become Solidly-Selected, and the other track will become Flashing-Selected. Now both Audio Tracks will operate as a paired (stereo) unit.

Hint: You can also use the Data Entry controls to select both Audio Tracks (L & R). ATRK-A (Audio Track A) will become Solidly-Selected, and ATRK-B (Audio Track B) will become Flashing-Selected.

Once you've selected the Record Source field values for your signal source, press either an ***Instrument*** button (the recommended way to select a destination Instrument) *or* press the ***Enter•Yes*** button. If the ***Enter•Yes*** button is pressed, the display will ask:

- Press an unused (LED not lit) ***Instrument•Sequence Track*** button. The ASR–10 goes into Level-Detect (or VU) mode:



In this mode, when recording in stereo, the VU meter will monitor the louder of the two channels signals. Once you have the level adjusted:

- Press ***Enter•Yes*** or press the Left Foot Switch to initiate sampling. The display will show WAITING... until the threshold is exceeded. The AMP annunciator will light if the Input Source clips. During sampling the display shows:

The xxx value (range 000 to 999) will count down in real time when sampling. Press any button to stop sampling. Remember that if the sampling threshold (the star) is set higher than the input signal level (the vertical bars), the ASR–10 will not begin sampling until the input signal crosses the threshold. Pressing ***Cancel•No*** while the sampling screen reads WAITING will abort sampling, and the ASR–10 will return to Level-Detect mode.

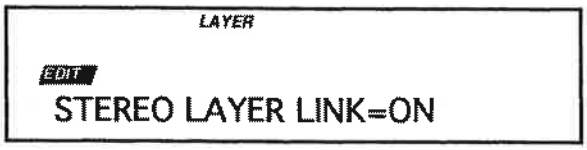
- Play the sound to be sampled; play the guitar, synth, start the recording, or speak into the microphone. If you are planning on using a microphone, we recommend plugging it into a mixer or preamp, and sending a line from the mixer/preamp to the Left Audio Input jack, for optimal performance.
- Press ***Cancel•No*** or press the Left Foot Switch to stop sampling. If you don't stop it by pressing ***Cancel•No*** or the foot switch, the ASR–10 will continue sampling until it has used up all the available memory.

After you have stopped sampling (or the memory is full) the display flashes PLAY ROOT KEY OR ENTER. The Root Key is the incoming MIDI note number that will play the WaveSample at its original pitch. In other words, if you play a C4 on the MIDI keyboard connected to the ASR–10, the sample will play back at "unity" (the same pitch as the original input signal) whenever the C4 key is pressed. Whichever key you press in response to the PLAY ROOT KEY OR ENTER prompt is the key that will play back exactly what you sampled. The samples you've just created are stored in companion layers.

Companion Layers

Stereo Samples are stored in pairs of adjacent *Layers* (1&2, 3&4, 5&6, or 7&8), with the LEFT channel always stored in an odd numbered Layers (1, 3, 5 or 7), and the RIGHT channel stored across the same Key-Range in the next higher even numbered Layer. These are called *companion* layers.

The Edit/Layer parameter STEREO LAYER LINK=xxx is automatically set to ON after a Stereo Sample is recorded. When this parameter is turned ON in the selected Layer, it is simultaneously turned ON in the companion layer. When STEREO LAYER LINK=ON, simultaneous editing of both channels of a Stereo Sample is enabled (see below).



The Stereo Layer Link parameter is located at Edit/Layer screen address "99," and can be quickly accessed by double-clicking the ***Layer*** button after pressing the ***Edit*** button.

- When STEREO LAYER LINK=ON, editing Layer or WaveSample parameters in one layer will edit the same parameters in the companion layer. Editing with the ***Data Entry Slider*** will set the current parameter in both layers to the same absolute value. Editing with the ***Up/Down Arrow*** buttons will edit parameter values in both layers relative to one another.
- When STEREO LAYER LINK=OFF, you can independently edit layers that contain one channel of a Stereo Sample.
- If the companion layer does not exist, STEREO LAYER LINK cannot be turned ON in the Selected Layer.

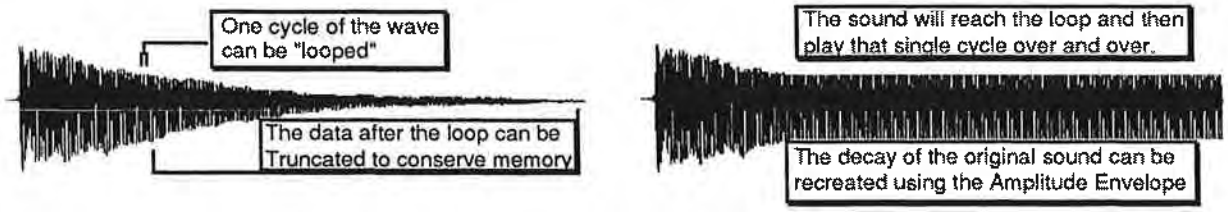
Looping

One of the fundamental challenges of sampling is to make efficient use of the sampler's memory. You simply cannot sample the entire duration of every event. Plus, you usually want a sound to continue playing as long as you're holding the key down, regardless of how long the original event was. This is why we employ *Looping*.

When we loop a segment of sound, what we are doing is playing through to the end of the looped segment and then instantly starting back at the beginning of the loop, as if playing one continuous sound. It's a lot like a tape loop — a segment of recording tape with the two ends spliced together so that it cycles repeatedly.

Consider that a low note on a grand piano can last 30 seconds or more from the time the key is struck to the time it decays into silence. Sampling this entire event does not make sense because:

- You could use up all your sampling memory on one note, leaving no memory for multisampling; and
- After the initial attack transients die down, the sound of the piano string settles into what is essentially a repeating waveform. This means that you can select one cycle of this repeating waveform and *loop* it (play it over and over). All the data after the loop can then be removed (truncated) because you don't need it. This conserves memory.


One cycle of the wave can be "looped"
The data after the loop can be Truncated to conserve memory
The sound will reach the loop and then play that single cycle over and over.
The decay of the original sound can be recreated using the Amplitude Envelope

A piano, solo violin, guitar, or woodwind are all sounds that you could loop in this way. However, there are other sounds that do not settle into a repeating waveform. A full pipe organ, string section or a choir would be good examples of such sounds. These sounds remain extremely complex throughout their duration. Using a single-cycle loop on a choir sample would cause the sound to suddenly become very static and one dimensional when it reaches the looped section. These complex, ever-changing sounds require that we employ longer loops — repeating a large segment of the sound rather than just a single wave cycle.

Generally, successful loops fall into one of two categories — single-cycle loops, which we refer to as short loops, and loops of about a second or more, which we call long loops. Which type to use depends on the sound. If the sound settles into something relatively static, use a short loop. If the sound is composed of many elements that constantly chorus and interact with each other over time, you will need to use a long loop.

Auto-Loop Finding

When you loop a sample, it is important that the beginning and end of the loop "match up" to form a continuous wave shape — otherwise the loop will pop or click each time it reaches the splice point. For this reason, the best loop points are those which occur where the wave crosses the zero line (called a "zero crossing") and where the direction of the wave at the loop start matches that at the loop end.

The ASR–10 has a system parameter called AUTO-LOOP FINDING that, when ON, allows you to choose only loop start and loop end points which meet these criteria.

Press ***Edit***, then ***System•MIDI***, and scroll until the display shows AUTO-LOOP FINDING = <ON,OFF>, and set it to ON. Now, whenever you move the loop start of any WaveSample, the ASR–10 will automatically jump to the next zero crossing which matches the wave direction at the loop end, and vice versa. This does not, in itself, guarantee a good loop but it will save you lots of time and trouble by ruling out all those splice points which definitely *won't* work.

Short (Single-Cycle) Loops

As discussed previously, a short (single-cycle) loop is appropriate when the sustain portion of the sound is a more or less repeating wave. The best way to find out if a short loop is right for a given sound is to try it and see if it works. If fact, the best way to learn about looping generally is to experiment with moving both the loop start and the loop end around in a WaveSample and listening to the results.

There is a simple technique for finding a short loop. For this example we will sample some sound with a simple sustain, such as a clean guitar or bass sound, or a synth patch without a lot of chorusing.

First press ***Edit***, then ***System•MIDI*** and make sure AUTO-LOOP FINDING=ON.

Next connect the signal source to the ASR–10 Audio In, set the input level, and sample the sound. After you have played the root key to conclude the sampling process, the ASR–10 puts you on the Edit/Wave page, which is just where you want to be to turn the loop on.

- Press the ***Up Arrow*** button twice to select MODE=LOOP FORWARD. Now hold down a key and listen to the sound. The default settings for a new WaveSample are LOOPSTART=00%, LOOPEND=99%, meaning that the whole sample is currently looping over and over.

- Scroll right to the LOOP START parameter, underlining the coarse adjust (the number to the right, in parentheses). Use the ***Data Entry Slider*** to move the loop start up, past the beginning of the sound. Try a setting of (25) to start. This puts the loop start 25% of the way into the WaveSample. Play the sound and listen.

- Scroll right to the LOOP END parameter, again underlining the coarse adjust (the number in parentheses) which should be at (99). Use the ***Data Entry Slider*** to move the loop end down as far as it can go. It will stop at (25) or wherever you set the loop start to, because you cannot adjust the loop end before the loop start. The loop start and loop end are now at the same place in the WaveSample. Play the sound. When it hits the loop, it will go supersonic because the ASR–10 is trying to loop on just one sample.

- Scroll two steps to the left to select the loop end fine adjust (the number just after LOOPEND=). Press the ***Up Arrow*** button once and play the sound. The pitch of the loop should be a little lower —you have moved the loop end away from the loop start to the first potentially good splice point.

- Press the ***Up Arrow*** button again and listen to the sound. With each press of the ***Up Arrow*** button, the pitch of the loop should go down as the ASR–10 moves the loop end to the next zero crossing. Keep pressing the ***Up Arrow*** button, one press at a time, until the loop sustains at the same pitch as the sound before the loop. This is a single-cycle loop.

- After you've found a good loop, you should move the sample end back to the same point as the loop end, then truncate the WaveSample to reclaim the memory that is no longer needed. (TRUNCATE WAVESAMPLE is found on the Command/Wave page.)

Adjusting Loop Position

Before truncating the WaveSample, you may decide that the loop occurs too early in the sound (before all the attack transients have died down) or too late (wasting memory). The ASR–10 has a way to move the whole loop (start and end) around while keeping its relative length intact.

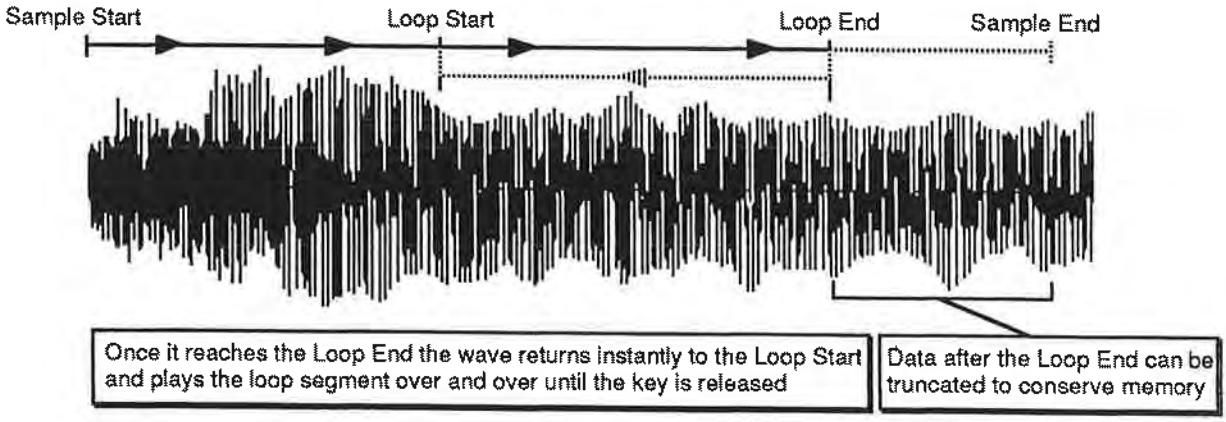
- While still on the Edit/Wave page (press ***Edit***, then ***Wave*** if you are not) scroll until the display shows LOOPPOS= ## (##). This is the loop position parameter. The number shown here is that of the loop start, but adjusting this parameter will move both loop start and loop end together, leaving the length (and thus the pitch) of the loop unaffected.

- Underline the loop position coarse adjust (the number in parentheses) and try moving the loop position around while listening to the sound. Find the place where the loop works most naturally with the rest of the sound. Note that you might have to readjust the loop end slightly after moving the loop position, particularly if the sound has pitch variations in it.

- Again, once you've found a loop you like, move the SAMPLE END back to the loop end, and truncate the WaveSample.

Long Loops

If a single-cycle loop doesn't work for a particular sound, try a longer loop. The idea is the same — a segment of the sound is repeated again and again, but the long loop encompasses many cycles of the sound. Long loops retain the tonal complexity and sense of "movement" in the original sound, and will therefore work better for sounds like vocals, strings, flutes, stacked synth sounds, etc.


Sample Start
Loop Start
Loop End
Sample End
Once it reaches the Loop End the wave returns instantly to the Loop Start and plays the loop segment over and over until the key is released
Data after the Loop End can be truncated to conserve memory

There is no one simple procedure for getting a perfect long loop every time — it all depends on the sound. Some sounds are easy to loop; other sounds seem to defy all attempts to find a good loop. Especially difficult to loop are those sounds whose timbre and volume change radically over time. You'll find, however, that the ASR–10 provides a wide variety of tools for getting the best possible results with even the most difficult sounds. By following the guidelines below (and not being afraid to experiment) you should be able to get an acceptable loop on virtually any sample.

- The AUTO-LOOP FINDING parameter should be on when looking for a long loop.
- Sample a sound, play the root key, and set the loop mode to MODE=LOOP FORWARD.
- Scroll to the LOOP START parameter, underlining the coarse adjust (in parentheses), and move the loop start forward past the initial attack of the sound. Try starting with a value somewhere between 15 and 40 percent. Hold down a key and listen to the loop.
- Scroll to the LOOP END parameter, again underlining the coarse adjust. While holding down a key, use the ***Data Entry Slider*** to move the loop end back towards the loop start. The closer to the loop start it gets (that is, the shorter the loop) the more noticeable the loop will sound. The idea is to find the shortest possible loop (to conserve memory) which doesn't sound obviously "looped."
- While holding down a key you can scroll back and forth between the LOOP START and LOOP END, changing the coarse and fine adjust for each, listening to how different parts of the sound loop.

- Most often you will find a loop that works fairly well except that the splice point is noticeable. In this case, try performing one (or more) of the crossfades contained on the Command/Wave page. Crossfading can smooth out the timbral differences between LOOP START and LOOP END, making the transition less abrupt.
- A loud bump or "pop" at the splice point usually indicates a volume difference between loop start and loop end. Try using the Command/Wave, VOLUME SMOOTHING command on the loop before doing a cross fade.
- Again, once you have a good loop, move the sample end back and then truncate the WaveSample.

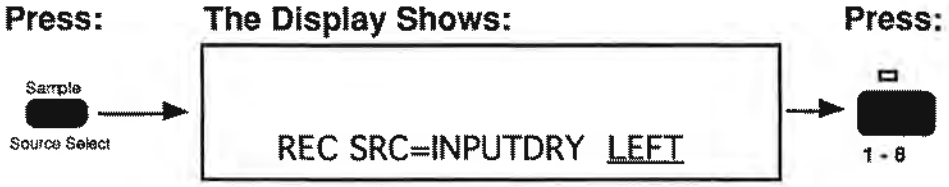
Sampling Signal Source Parameters

Section 8 — Sampling/Signal Source Parameters

This section describes the sampling and signal source parameters. For a general overview of the concepts involved, refer to the previous section.

Sampling Parameters

There are a number of sampling parameters you can adjust. These parameters are accessed by pressing the ***Left/Right Arrow*** buttons while in Level-Detect mode. To enter Level-Detect mode:

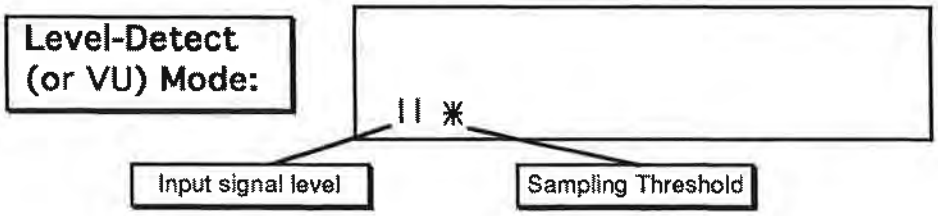


- Press the ***Sample•Source Select*** button to the right of the display. This is the Record Source parameter screen. The RECORD SOURCE parameter has 2 fields. Field 1 determines which Audio Source will be sampled. Field 2 determines whether the Audio Source will be sampled in stereo or mono. For more information about these fields, see *Section 7 — Sampling/Signal Source Concepts.*

- Press an unused (no LED lit) ***Instrument•Sequence Track*** button.

Sample Inst•Track	**SAMPLING THRESHOLD LEVEL**
	Press Sample•Source Select / Instrument•Sequence Track

After you've pressed an unused ***Instrument•Sequence Track*** button, the ASR–10 goes into Level-Detect (or VU) mode:



In Level-Detect mode, the display acts like a peak-reading VU meter. The vertical bars on the display lighting from left to right show the level of the incoming signal. The *star* represents the sampling threshold — the ASR–10 will not begin recording until the signal crosses this level. The Sampling Threshold provides a way to make the ASR–10 "wait" until the signal reaches a certain level before it begins to record. The ASR–10 offers 21 possible sample threshold levels.

To Adjust the Sampling Threshold Level:

- Press the ***Up Arrow*** button to raise the threshold, or the ***Down Arrow*** button to lower it.

When the sampling threshold is set to minimum (the star all the way to the left of the display) the ASR–10 will begin sampling as soon as you hit ***Enter•Yes***. The higher (further to the right) you set the threshold, the higher the input signal will have to be before the ASR–10 begins recording.

Hint: While in Level-Detect Mode, you can select, deselect and stack (double-click) ***Instrument•Sequence Tracks***, and can play them from an external MIDI controller, even during sampling. You can also press ***Play*** and ***Stop•Continue*** to sample along with sequence playback, or even to sample sequence playback when REC SRC= MAIN-OUT. This way, what you are about to sample can "sync-up" to what you've already sequenced or sampled (pitch, timing, etc.). If you scroll right or left to view any of the other sampling screens, you will only hear the destination instrument (the instrument into which you are sampling).

Hint: By adjusting the sampling threshold level, you can avoid recording a lot of empty space (silence) at the beginning of your sample.

Sample Inst•Track	**SAMPLE RATE**
	Press Sample / Instrument•Track / Scroll using the Left/Right Arrow buttons

This read-only parameter shows the current system sample rate, as determined by the current effect on the FX Select•FX Bypass page. If you attempt to edit this parameter using the Data Entry Controls, the display shows either CURRENT EFFECT IS 30K or CURRENT EFFECT IS 44K.

Read-only Range: 29.7619 or 44.1000 KHZ

Now press the ***Right Arrow*** button again.

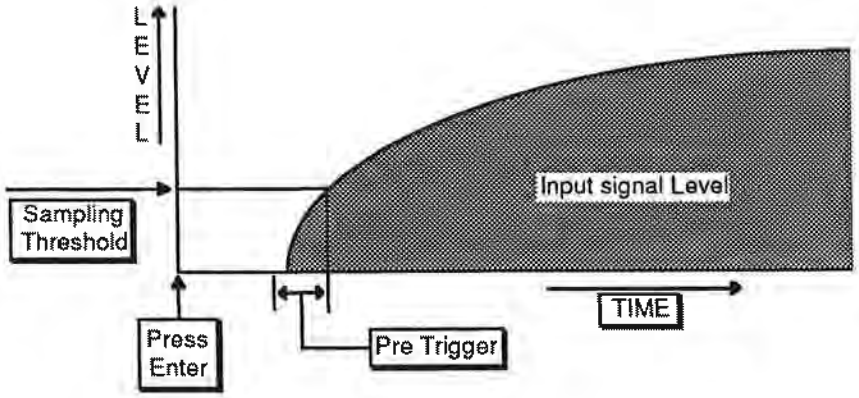
Sample Inst•Track	**PRE TRIGGER**
	Press Sample / Instrument•Track / Scroll using the Left/Right Arrow buttons

Range: 0 to 140 ms

This lets the ASR–10 start the sample *before* the sampling threshold is reached. The problem with using a high sampling threshold is that you can sometimes chop off low-level information at the beginning of the sound (below the threshold), making it sound clipped and unnatural. That's where the PRE TRIGGER parameter comes in.

The ASR–10 initially records the sample into a "buffer memory" of up to 140 milliseconds. You determine the length of this buffer with the pre trigger amount. Now when you press ***Enter•Yes***, you get the benefits of being able to use a higher threshold without chopping off the beginning of the sound.

After the sampling threshold is reached, the ASR–10 will "back up" the SAMPLE START by the amount of the time selected in the PRE TRIGGER parameter. This allows sound just below the threshold level (which would otherwise be lost) to be retained, as shown below:



A pre-trigger amount of 10 to 20 ms should be adequate in most cases. You could set it higher to ensure that you don't miss any of the sound, or when using a very high sample threshold. In any case, a little silence in front of the sound is better than clipping it short, because you can later move the sample start forward to eliminate the dead space, and then truncate the WaveSample.

- Press the ***Right Arrow*** button again.

Sample Inst•Track	**SAMPLE TIME**
	Press Sample / Instrument•Track / Scroll using the Left/Right Arrow buttons

Range: Various

This tells you how much sampling time is available, taking into account the current system sample rate and the amount of free Internal Memory. This is a read-only parameter and can not be adjusted (though changing the system sample rate on the FX Select•FX Bypass page will cause this number to change).

- Pressing the ***Right Arrow*** button once more returns you to the Level-Detect mode.

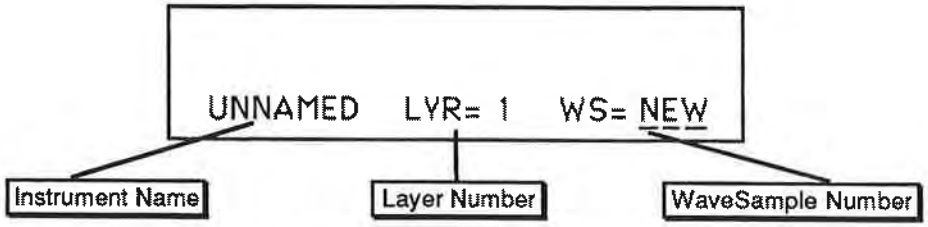
Additional Multisampling Parameters

When you begin multisampling (making several samples within one instrument), there are three additional parameters that appear on the multisampling screen. These parameters allow you to record a new (additional) sample, sample over (replace) an already existing sample, or select a new/different layer within an already existing instrument.

Once you've recorded your first sample and you want to make a second sample within the same Instrument:

- Press the ***Sample•Source Select*** button.
- Press the ***Instrument•Sequence Track*** button of the instrument you just sampled.

When you select an existing instrument from the REC SRC screen, the following multisampling screen (also called the Sampling Context screen) appears:



The multisampling parameters are:

- Instrument Name — The first eight letters of the instrument name (which is always UNNAMED for newly sampled instruments) are shown. This is a read-only parameter and cannot be adjusted.

- Layer Number — This parameter shows the number of the current layer. Use the ***Left/Right Arrow*** buttons to select this parameter, and the ***Up/Down Arrows*** to choose an already existing layer, or a new layer (LYR=N). Remember, each instrument can have up to eight different layers.

  If you've recorded a stereo sample, the layer parameter will show either an **L** or an **R** following the layer number, indicating that they contain the **L**eft or **R**ight channels of a stereo sample.

- WaveSample Number — This parameter determines which WaveSample will be sampled after you press ***Enter•Yes***. The default here is NEW, meaning that a new WaveSample will be created. You can however, select an existing WaveSample to sample over by using the ***Up/Down Arrow*** buttons, the ***Data Entry Slider***, or by pressing a key on the keyboard.

  When you sample over an existing WaveSample, the ASR–10 will preserve the keyboard range, all WaveSample parameters (envelopes, etc.) and, when possible, the loop points of the original sample.

Note: When an illegal layer (such as an existing mono layer) is chosen as the destination for stereo samples, the following message will be briefly displayed:

If no legal companion layers are available in the selected Instrument, the display will also show the above screen.

Select a legal layer and press ***Enter•*Yes** to return to Level-Detect mode.

WaveSample & Layer Concepts

Section 9 — WaveSample and Layer Concepts

This section provides an overview of the concepts behind editing an ASR–10 WaveSample. For more detailed descriptions of the actual parameters, refer to the following section.

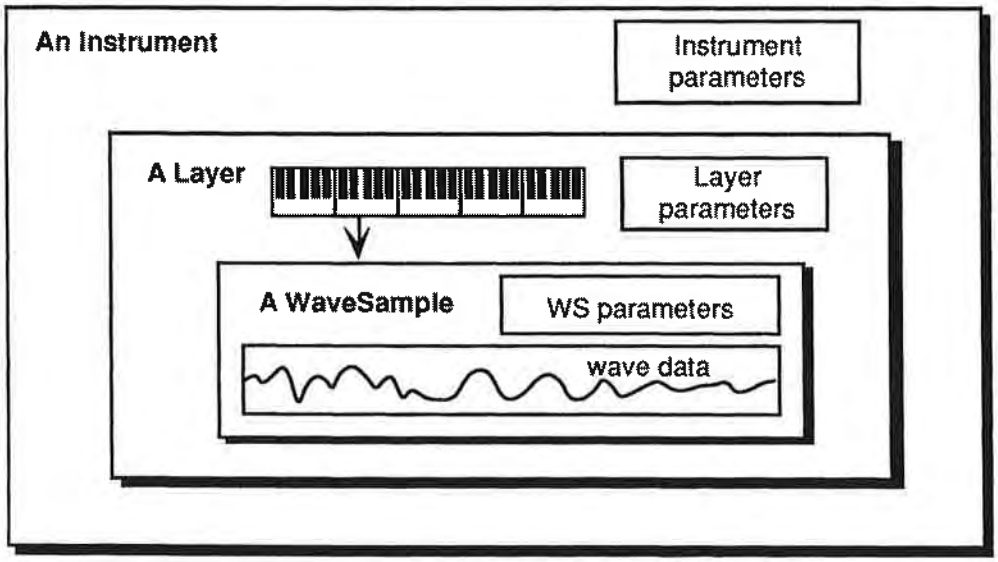
ASR–10 Instrument Configuration

Each ASR–10 instrument consists of up to eight Layers, and up to 127 WaveSamples. Each WaveSample within an ASR–10 instrument consists of the following:

- a digital oscillator playing a wave from memory
- two multi-mode digital filters
- one LFO (Low Frequency Oscillator)
- one random NOISE generator
- three complex envelope generators for controlling pitch, filter, and volume
- a versatile matrix modulation scheme with 15 routable modulation sources

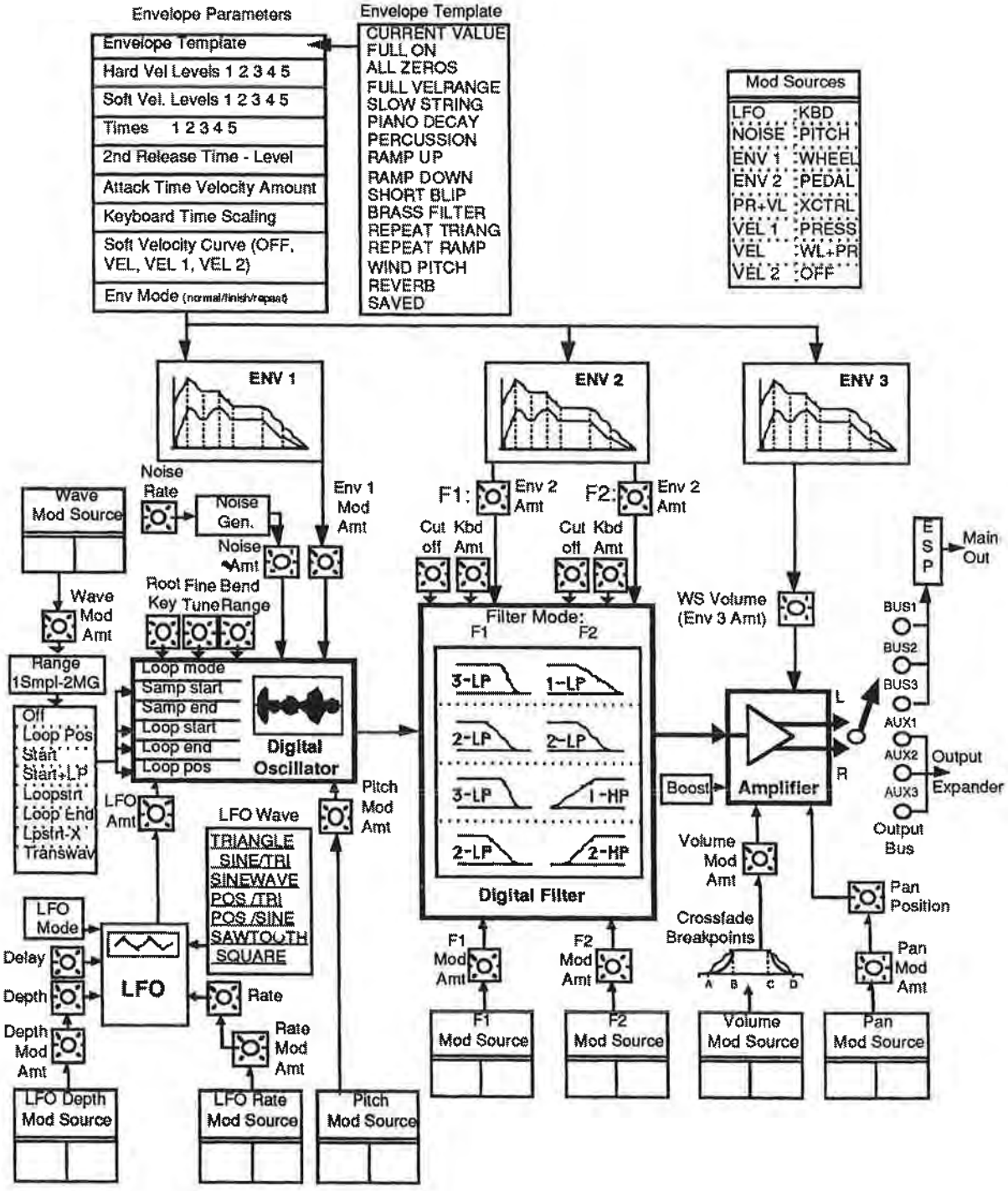
About WaveSamples

The WaveSample is the most basic unit of sound within the ASR–10. A WaveSample can be anything — a drum, a note on a piano, a sound effect, or your voice. Each WaveSample has its own set of WaveSample parameter settings. Several WaveSamples can be grouped together to form *Layers*, and up to eight layers can reside within an instrument.



The diagram on the following page shows the configuration of one ASR–10 WaveSample.

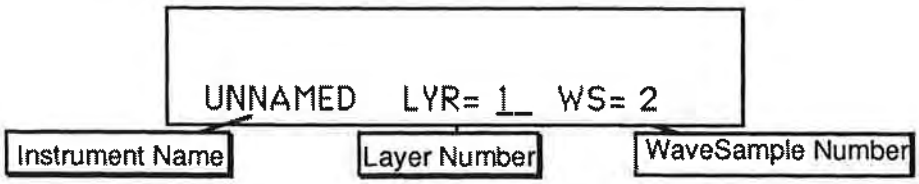
ASR–10 WaveSample Configuration



Selecting a WaveSample or Layer for Editing (the Edit Context page)

In the next two sections of the manual, all the tools available for shaping a raw WaveSample will be covered. Almost all of these parameters and commands will require you to select either a WaveSample or layer for editing. The procedure for doing this is quite simple:

- Press the ***Instrument•Sequence Track*** button for the instrument you want to edit.
- Press the ***Edit*** button. The ASR–10 displays the Edit Context page:



You see the currently selected instrument, layer (stereo or mono), and WaveSample.

- Press the ***Left/Right Arrow*** buttons to move the cursor (underline) until it is beneath the WaveSample Number (WS=__).
- Play the MIDI keyboard connected to the ASR–10 to select the WaveSample you want to work on. When you play the keyboard, the WaveSample Number showing on the display will change as you play different WaveSamples. Whichever WaveSample Number is showing is considered the current "edit WaveSample" — the one that will be affected by any WaveSample Edit or Command functions.
- Pressing ***Edit*** again will return you to the last Edit page you were on. This lets you select a different WaveSample and then quickly return to the same parameter (SAMPLE START or END, for example) that you were editing.
- To perform editing on all WaveSamples in an entire layer, make sure the cursor is beneath the Layer Number (LYR=__). The WaveSample Number reads WS=ALL. You can select the various layers within the instrument by using the ***Data Entry Controls***. Only layers active in the current patch can be edited (active layers are displayed on the Edit/Instrument PATCH= parameter). If the layer number has an "L" or an "R" directly after it, then it is defined as a Left or Right stereo layer. Any edits to WaveSamples in a stereo layer will also affect the WaveSamples in its companion layer, even if the companion layer is not active in the current patch.
- To perform editing on all active layers and WaveSamples in an instrument, make sure the cursor is beneath the Instrument Name. The Layer Number reads LYR=A. Now any WaveSample parameter edits will affect all layers and WaveSamples in the current patch.
- When LYR=A or WS=ALL, the ***Data Entry Controls*** function as described below:

Note: When STEREO LAYER LINK= ON, editing layer or WaveSample parameters in one layer will edit the same parameter in the companion layer. Editing with the ***Data Entry Slider*** will set the current parameter in both layers to the same absolute value. Editing with the ***Up/Down Arrow*** buttons will edit parameter values in both layers *relative* to one another.

Hint: When the cursor is beneath the Layer Number on the Edit Context page, selecting any WaveSample Edit page, and pressing different keys, will display the parameter values for the WaveSamples assigned to that key range in the selected layer.

Hint: When editing any WaveSample parameter, pressing ***Cancel•No*** will reset the parameter to the value it was set to before it was edited.

WaveSample Commands

There are many commands for working with WaveSamples and wave data. Rather than putting them all on one page, they are broken down into three pages.

The Command/Wave page contains the standard copy and delete commands for WaveSamples, plus several looping commands.

The Command/Amp and Command/LFO pages contain amplitude and wave data processing commands respectively. Some of these are fairly low-level operations that are rarely used. They are included for completeness.

Note: Always make sure the WaveSample you want to edit is selected on the Edit page.

Messages

- NO EDIT WS SELECTED — go to the Edit page and select a WaveSample you want to work on.
- WAVESAMPLE IS A COPY — means the WaveSample you are editing is not an original, but a copy. If you press ***Enter•Yes*** in response to this message, the ASR–10 will take the data from the original WaveSample and copy it into the WaveSample you are editing. This will make the instrument you are editing larger. If this is undesirable, use the WAVESAMPLE INFORMATION command to find which WaveSample is the original. Then perform the command on the original.
- LAYER NOT IN PATCH — press ***Edit***, then ***Instrument*** and select the correct layer. Use the ***Arrow*** buttons and enable the layer in the current patch.
- SAMPLES NOT SAME SIZE — means that the WaveSample is in a stereo layer, and its companion WaveSample is not the same size. This message is designed to prevent accidental damage to mono WaveSamples in stereo layers. If you're sure that you want to perform the command, press Edit/Layer and set STEREO LAYER LINK= OFF.
- NO AUDITION - OK? — when there is not enough free memory available for the Audition function to operate, this screen will appear. This is because the Audition function requires as much free memory as the size of the sample being processed. When processing samples in stereo layers, the Audition function requires *twice* as much free memory. If you press ***Enter•Yes*** in response to this message, the ASR–10 will execute the command *without* any audition. Pressing ***Cancel•No*** will abort the command.

Note: Wherever you see the notation ##, that symbol will represent the numeric value of a parameter whose setting or level is considered arbitrary to the subsequent discussion. For example, if SCALE DEPTH= ## DB, the ## represents the entire range of this parameter. You can use the ***Data Entry Slider*** or ***Up/Down Arrow*** buttons to explore the range of values.

WaveSample Commands in Stereo Layers

When Edit/Layer STEREO LAYER LINK= ON, the wave data commands on the Command/Wave, Command/Amp, and Command/LFO pages will be simultaneously performed on the corresponding WaveSample in the companion layer.

Modulators

About Modulation

To modulate something is simply to cause it to change. Within the voice architecture of the ASR–10 we begin by setting basic, or manual, levels for the pitch, brightness, and volume of a voice, and we then modulate those levels in various ways to create movement and dynamics.

Suppose you switch on your stereo, and turn the volume half way up. We can call this the manual volume setting. It will stay at that level until it's changed. Now suppose that you take the volume knob of your stereo and begin quickly turning it up and down, so the volume gets continuously louder and softer, louder and softer. What you would be doing is modulating the volume of your stereo. Your hand would be the *Modulation Source* and volume would be the *Modulation Destination*. If you were to take the treble control, and do the same to that knob, you would be modulating the brightness of your stereo. In this case, brightness would be the *Modulation Destination*.

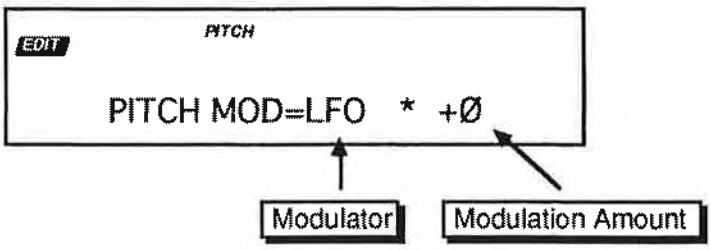
In much the same way we modulate various levels within the ASR–10 (though generally the approach is less haphazard). There are 15 different *Modulation Sources* (or modulators) available, and they can each be independently assigned to vary the manual levels for a great many aspects of a voice, including real time control of aspects of the current effects algorithm. The parameter that will be modulated by the modulator is called the *Modulation Destination*. In a stereo layer, the right channel will always use the modulation sources and destinations of the left channel.

Selecting a Modulator

On those programming pages where a modulator can be selected to modulate a given modulation destination, the display shows MOD=__ (short for Modulation Source). A modulator is chosen by selecting (underlining) the MOD= parameter with the ***Left/Right Arrow*** buttons, and then using the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to select among the 15 available modulation sources.

Hint: Moving the ***Data Entry Slider*** all the way up selects OFF, which is handy if you don't want a modulator applied to a particular destination.

Let's take, for example, the Pitch Mod parameter, which is where you apply modulation to the pitch of a WaveSample (i.e. Pitch is the modulation destination). Press ***Edit***, then ***Pitch***. Next, press the ***Right Arrow*** button repeatedly until the following display appears:



Modulation Amount

As shown above, where a modulation source is selected, the parameter immediately to its right controls the *Modulation Amount* which controls how deeply the selected modulator will affect the level to which it is being applied.

Press the ***Right Arrow*** button to underline the Modulation Amount, and use the data entry controls to adjust it. Modulation amount can be positive or negative. A modulation amount of +0 has the same effect as turning the modulator OFF.

Modulation Sources

The 15 Modulation Sources available on the ASR–10 are as follows:

- LFO - Low Frequency Oscillator
- NOISE - Noise Generator
- ENV 1 - Envelope 1
- ENV 2 - Envelope 2
- PR+VL - Pressure + Velocity
- VEL 1 - Convex Velocity Curve
- VEL - Linear Velocity Curve
- VEL 2 - Concave Velocity Curve
- KBD - Keyboard Tracking
- PITCH - Pitch Bend Wheel
- WHEEL - Modulation Wheel
- PEDAL - Voltage Control Foot Pedal
- XCTRL - External Controller (MIDI)
- PRESS - Pressure (After-touch)
- WL+PR - Mod Wheel + Pressure
- OFF - No Mod Source selected

- **LFO — Low Frequency Oscillator**

  The *Low Frequency Oscillator* generates only very low frequency waves below the audio spectrum, which can produce vibrato, tremolo, and many other effects, depending on the LFO wave selected, and where it is applied as a modulator. There are seven possible waveshapes for the LFO.

- **NOISE — Noise Generator**

  The Noise Generator produces a randomly changing level. It is useful for modulating, among other things, the pitch of a WaveSample (Edit/Pitch page). Applied to pitch with large modulation amounts it tends to create strange "computer sound" effects. Small modulation amounts (around +2 to +4) can create a subtle random movement in the sound, which can impart a more natural quality.

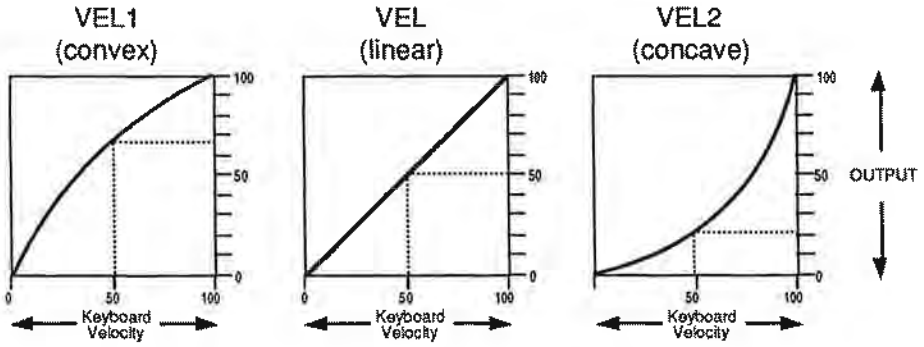
  When STEREO LAYER LINK= ON, the NOISE modulation source in the right layer will be synchronized with the NOISE modulation source in WaveSamples assigned to the same or overlapping key zones in the left layer.

- **ENV 1, ENV 2**

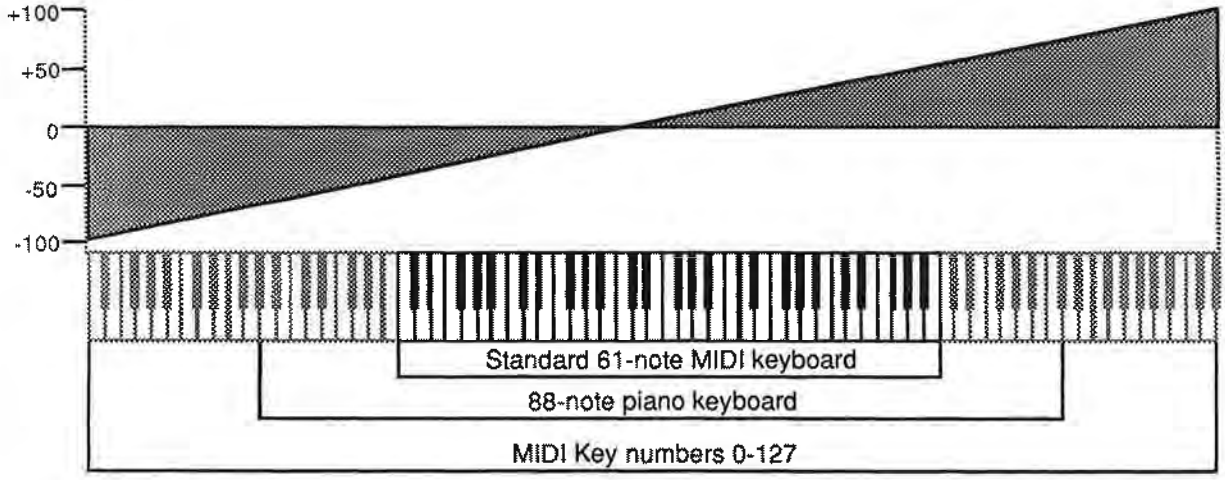
  The ASR–10 has three complex *Envelopes*. Envelopes are used to create changes over time in brightness, pitch, volume, etc.
  - ENV 1 is permanently routed to the pitch of the WaveSample, though it can be assigned as modulator elsewhere if you wish.
  - ENV 2 is permanently routed to the filter cutoff frequency. It also can be assigned as a modulator elsewhere.
  - ENV 3 is a special case. ENV 3 *always* controls the volume of the WaveSample, and cannot be selected as a modulator.

- **PR+VL — Keyboard Pressure + Velocity**
  This is one of two "combination" modulators. When this modulator is selected, both pressure *and* velocity will affect the modulation destination. After the note-on velocity has modulated the note played, pressing into the keyboard will change the modulator output set by velocity. After pressure has been exerted on the keyboard, pressure becomes the modulator until the note is released.

- **VEL 1, VEL, VEL 2 — Velocity**
  Velocity means how hard you strike a key. Selecting VEL 1, VEL, or VEL 2 as a modulator allows you to modulate any manual level with velocity. Velocity as a modulation source only goes positive (though assigning a negative modulation amount will make the net result be to reduce the level with increased velocity). Each velocity modulator has its own response curve, shown below:



- **KBD — Keyboard Tracking**
  This uses the position of a note on the keyboard as a modulator. The scaling effect of this modulator is figured over the full 128 MIDI key numbers:



As the above illustration shows, the effect on KBD as the modulator goes negative as well as positive. The effect of KBD is to reduce the manual level of notes below the break point (middle C, C4, MIDI Note #60), and increase the manual level on notes above the break point. Negative modulation amounts will do the opposite.

- **PITCH — Pitch Bend Wheel**
  This complex modulator groups the *Pitch Bend Wheel*, along with any other pitch modulators and pitch tables assigned to the instrument, to form a combined modulator. PITCH takes the amounts of each individual modulation source (as well as any pitch table information assigned to the instrument) and adds them together, taking the sum total as the modulation amount. Applied to the filter cutoff frequency for example, this would cause notes to become brighter as you bend them upwards and more muted as you bend them down (or the opposite with negative modulation amounts), in addition to the effect of any other filter cutoff modulators. If a pitch table was employed, the brightness would increase and decrease with the individual pitch assigned to each note, in addition to the effect of the pitch wheel and any other modulators. The value of the pitch bend wheel is 0 to 99, normally at the center (50% at rest).

- **WHEEL — Modulation Wheel**
  The *Mod Wheel* to the left of the keyboard is assignable wherever a modulator is selected. To use the mod wheel for vibrato (one common application), WHEEL must be assigned to modulate the LFO, and the LFO Amount set to some number other than zero on the Edit/Pitch page. The mod wheel's effect is positive-going only, from 0 (wheel towards you) to +99 (wheel away from you). Negative modulation amounts will reverse the effect.

- **PEDAL — Voltage Control Foot Pedal**
  This selects the optional *CVP-1 Foot Pedal*, which can be plugged into the Pedal•CV jack on the ASR–10 rear panel, as a modulator. Its effect is positive-going only, like the mod wheel. It can be applied wherever a modulator is selected.

  Note that the Foot Pedal will only act as a modulator when the Pedal Function parameter is set to PEDAL=MOD on the Edit/System•MIDI page. When that parameter is set to PEDAL=VOL, the Foot Pedal will act as a volume pedal, not as a modulator (though this has no effect on incoming MIDI Foot Pedal data).

- **XCTRL — External Controller (MIDI only)**
  An external controller (such as a Breath Controller, etc.) that is received via MIDI from another synthesizer or controller, can be assigned as a modulator within ASR–10 instruments. On the Edit/System•MIDI page, you can select the number of the external controller that will be recognized by the ASR–10.

Hint: You don't have to be playing the ASR–10 from an external instrument for this to work. For example, if you have a keyboard with a Breath Controller, do the following:

- Connect the breath controller's MIDI Out to the ASR–10 MIDI In.
- Make sure both instruments have controllers enabled (Edit/System•MIDI page).
- Select Breath Controller as the external controller that will be received by the ASR–10 (XCTRL=02, also on the Edit/System•MIDI page).
- Assign XCTRL as a modulator for LFO level, Filter Cutoff frequency, or some other manual level within a WaveSample.
- Play the sound from the controller keyboard, while blowing into the Breath Controller connected to the sending instrument. The modulation will have the same effect as if you were playing from the sending instrument.

- **PRESS — Pressure (After-touch)**
  Pressure, also called after-touch, is a modulator that varies a manual level within a voice depending on how hard you press down on a key or keys. After you have struck a key, and while the note is sustaining, continuing to press down harder on the key brings in pressure. The ASR–10 receives pressure, and by using this modulator you can add a tremendous amount of expression to your sounds without ever taking your hands off the controller keyboard. Pressure can be assigned to alter the pitch or volume of voices, filter cutoff frequency, LFO rate or depth, pan location, etc.

There are two types of Pressure:

- Channel Pressure, also called Mono pressure, affects all notes that are playing when you exert pressure on any of the keys. For example, if you play a three note chord, pressing down harder on any of the three notes of the chord will modulate *all three notes*. This type of pressure is the more common of the two types.

  Most MIDI instruments that currently implement pressure send and receive only channel pressure. If you are playing such an instrument from the ASR–10, you should set the ASR–10 to send channel pressure. (Note that some devices, including *all* ENSONIQ products, transmit and respond to both types of pressure.)

- Poly-Key™ Pressure, also referred to as polyphonic pressure, is a more sophisticated and expressive type of pressure. Poly-Key pressure affects each key independently. For example, if you play a three-note chord, pressing down harder on any of the three notes of the chord will modulate *only that note*. The other two notes will remain unaffected. Most ENSONIQ keyboards transmit Poly-Key pressure, allowing you to use this expressive feature on your ASR–120 Module.

  Each preset or sequencer track can be programmed to generate Poly-Key pressure, channel pressure or none at all. If you wish to change the pressure type for a given track, you can do so on the Edit/Instrument page.

Hint: Poly-Key pressure generates a tremendous amount of data and will consume sequencer memory much faster than other types of events, such as notes and program changes. You should turn pressure off when sequencing instruments which do not respond to pressure, such as piano and drum sounds.

The PRESSURE MODE parameter on the Edit/Instrument page determines which of the two types of pressure will be used by the instrument. When set to KEY or CHAN, the ASR–10 will generate the selected pressure type, and will receive both types via MIDI. This parameter can also be set to OFF, in which case the instrument will not transmit or receive pressure via MIDI.

Note that not all instruments are necessarily programmed to respond to pressure. If pressure seems to have no effect when you play certain instruments, it is likely that the programmer did not assign pressure as a modulator anywhere within the instrument.

The effect of pressure as a modulator is positive going only, though assigning a negative modulation depth will cause increased pressure to reduce manual levels.

- **WL+PR — Wheel + Pressure**
  Another combination modulator. When selected, both the mod wheel *and* pressure will affect the modulation destination. This can be good for modulating LFO depth when using the LFO for vibrato. Either the Mod Wheel or Pressure can then be used to generate vibrato.

- **OFF**
  Modulation is disabled.

Wave Data Parameters

Section 10 — Wave Data Parameters

This section covers the parameters and commands that relate directly to the wave data, including sample length, playback direction, volume commands, and looping. For an overview of the WaveSample Programming concepts, refer to the previous section.

Edit/Wave Page

The parameters on this page control all the WaveSample pointers such as sample start, sample end, loop points, etc. Also on this page, you can choose the playback mode and set up modulation of the sample start and/or loop position.

EDIT WAVE	**MODE (WS Playback Mode)**
	Press Edit / Wave / 0

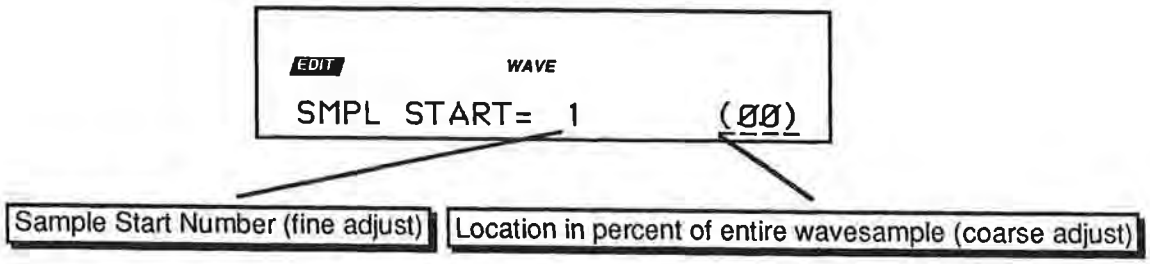
This parameter allows you to select one of five possible WaveSample playback modes.

- FORWARD-NO LOOP — The ASR–10 plays through the WaveSample once, from sample start to sample end.
- BACKWARD-NO LOOP — The ASR–10 plays through the WaveSample once, only in reverse, from sample end to sample start.
- LOOP FORWARD — The ASR–10 will play forward from sample start to loop end and return to loop start and will continue playing the loop while the key is depressed. Once the key is released the loop continues playing as the WaveSample envelopes away.
- LOOP BIDIRECTION — The ASR–10 will play forward from sample start. At loop end, the ASR–10 will play the WaveSample in reverse to the loop start, playing the loop section back and forth until the key is released.
- LOOP AND RELEASE — The ASR–10 will play forward from sample start and will continue playing the loop while the key is depressed. As soon as the key is released the ASR–10 will go into release mode from wherever it is and will keep playing until the end of the WaveSample, playing data after the loop. LOOP AND RELEASE works in conjunction with Envelope 3, which needs to have its MODE parameter set to CYCLE and its time and level values adjusted properly.

EDIT WAVE	**SMPL START**
	Press Edit / Wave / 1

This parameter allows you to set the sample start point. To conserve memory, it's a good idea to move the Sample Start and Sample End to eliminate any "dead space" at beginning or end, and then truncate the WaveSample using the TRUNCATE WAVESAMPLE command on the Command/Wave page (later in this section).

When the SAMPLE START or SAMPLE END parameters are displayed on the Edit/Wave page there are two ways to adjust them — you can move in large steps by adjusting the percent setting, or move one sample at a time by choosing the fine adjust.



The Coarse Adjust reading is the one in parentheses. Changing this value will move the Sample Start in steps of one percent of the entire WaveSample. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to move the Sample Start ahead until the dead space is gone when you play.

Once you have gotten close using the percent setting, press the ***Left Arrow*** button to underline the Fine Adjust setting. Now the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons will move the SAMPLE START in small steps (less than 1%), letting you fine-tune its exact position while playing the sample.

EDIT WAVE	**SAMPLE END**
	Press Edit / Wave / 2

This parameter allows you to set the sample end point. Again you see the sample number (Fine Adjust) and the percent (Coarse Adjust). Use the data entry controls to edit this parameter in the same fashion as the SAMPLE START parameter.

EDIT WAVE	**LOOPSTART**
	Press Edit / Wave / 3

This parameter allows you to set the loop start point. After making a sample it is often necessary to "loop" (play over and over) a portion of the sound, so that it can sustain indefinitely when the key is held down. Getting a good loop tends to be the most difficult part of sampling.

Here are a few steps to get you started once you have sampled a sound that you want to loop:

- Turn ON the AUTO-LOOP FINDING parameter on the Edit/System•MIDI page: Press ***Edit***, then ***System•MIDI***, then scroll left to the parameter AUTO-LOOP FINDING. Make sure it is ON. When AUTO-LOOP FINDING=ON, every time you move the LOOP START or LOOP END points of a WaveSample, the ASR–10 picks only "optimized" loop points – ones most likely to work.
- On the Edit/Wave page, the MODE parameter determines whether the WaveSample will loop (among other things). Set it to MODE=LOOP FORWARD.
- Select the LOOP START parameter.
  Move the cursor to underline the Loop Start Percent value (the number in parentheses). While holding down a note, move it away from the beginning of the WaveSample (the 25% to 50% range is usually a good starting point).

LOOP START is used in conjunction with the next parameter.

EDIT WAVE	**LOOPEND**
	Press Edit / Wave / 4

This parameter allows you to set or change the loop end point.

Move the cursor to underline the Loop End Percent value (in parentheses) and move it towards the beginning of the WaveSample, using the data entry controls to adjust in the same fashion as the LOOP START parameter. Move the LOOP START and the LOOP END around until you find the best sounding loop.

Note: The maximum loop length is 1.5 MegaWords.

Hint: There is a quick way to copy any of the SAMPLE START, SAMPLE END, LOOP START, and LOOP END parameter values from one parameter to another. This can eliminate the hassle of having to write the value down when, for example, you want the SAMPLE START to be set to the same value as the LOOP START, or vice versa. To use this function:

- Underline the value to be copied.
- Press the ***Enter•Yes*** button.
- Go to the parameter that you want to paste the copied value into (make sure this destination parameter value is underlined).
- Press the ***Cancel•No*** button.

EDIT WAVE	LOOPPOS
	Press Edit / Wave / 5

This parameter displays the LOOP START but moves both LOOP START and LOOP END, keeping the size of the loop the same. It lets you search for the perfect loop location after you've set up an optimal loop length.

EDIT WAVE	MOD / SRC
	Press Edit / Wave / 6 (Mod Type) 7 (Mod Source)

These parameters allow you to modulate certain parameters of the wave and select the mod source. The first parameter is WAVE MOD TYPE. This specifies which of the wave parameters the modulation will be affecting. There are seven possible MOD TYPES:

- OFF — Wave modulation is disabled.
- LOOP POS — Selecting this setting will modulate the loop position of the wave, moving it forward (towards the end of the wave) with positive values and backward (toward the beginning of the wave) with negative values.
- START — The SMPL START will be modulated.
- START+LP — Both the SMPL START *and* the Loop Position will be modulated. When START+LP is selected, sample playback will always begin at the LOOP START value, rather than at the SMPL START value.
- LOOPSTRT — Modulates the Loop Start parameter. This will have the effect of either making the loop shorter (by using positive modulation values to move the Loop Start point closer to the Loop End) or longer (negative values moving the Loop Start away from the Loop End).
- LOOP END — Works the same as LOOPSTRT, but from the back end.
- LPSTRT–X — This setting works the same as LOOPSTRT, but a pitch correction scheme is now employed to keep the pitch constant as the loop gets longer and shorter.
- TRANSWAV — Similar to LOOP POS, but the loop will move in increments of exactly one loop length. For example, with LOOPSTART at 0 and LOOPEND at 100, positive modulation will cause the loop to jump to LOOPSTART 100, LOOPEND 200, and then LOOPSTART 200, LOOPEND 300, etc. When this setting is selected, the MOD RANGE parameter (see below) has no effect, but the MOD AMOUNT (also below) sets the number of "jumps" that will happen over the full range of the controller (MOD SRC).

Any of the 15 ASR–10 modulation sources can be used as the SRC (Modulation Source) for the wave modulation.

EDIT WAVE	**MOD AMT / RANGE**
	Press Edit / Wave / 8 (Mod Amount) 9 (Mod Range)

Use these parameters to set the amount of wave modulation selected with the previous parameter. The wave modulation amount has two components:

- MOD AMT — This determines the amount of modulation that will be added to the wave parameter specified in the WAVE MOD TYPE parameter. This parameter has a value range of -99 to +99, allowing both positive and negative modulation amounts.
- RANGE — This portion of the parameter determines range that will be covered by the MOD AMT. This range is displayed in sample size. The reason for this is due to the nature of sampled instruments. Where on a synthesizer you would have waves or oscillators that had a defined length that was always the same, with samplers the wave size is always different from instrument to instrument, due to the sample time and rate at which each was sampled. The MOD AMT has no way of knowing this sample size, so the ASR–10 uses the RANGE to tell it.

Range: 1 Sample to 2 MegaSamples

To determine what size to set the RANGE parameter, go to the SAMPLE END parameter and look at the Sample End number. This number tells you how large the selected sample is in size. You can then select the RANGE value that comes closest to the actual size of the sample.

If you were working with a very small sample you would adjust the wave modulation amount to something like this MOD AMT=+99 RANGE=128. If you were working with a much larger sample you would adjust the wave modulation to MOD AMT=+99 RANGE=1 MG (MegaSample). Think of RANGE as a multiplier, and MOD AMT=-99 TO +99 as fine tuning.

Command/Wave Page

CMD WAVE	CREATE NEW WAVESAMPLE
	Press Command / Wave / 0

A primary function of the ASR–10 is to create new WaveSamples by sampling external sound sources. However, the ASR–10 also can create a WaveSample without sampling. The CREATE NEW WAVESAMPLE command creates a small square-wave without sampling. This is useful as a tuning reference, or for experimenting with voice parameters on a static wave.

CMD WAVE	COPY WAVESAMPLE
	Press Command / Wave / 1

Use this command to make a copy of the currently selected WaveSample and its data, or a copy of its parameters only (when used within an instrument).

COPY WAVESAMPLE lets you apply different WaveSample parameter settings to the same wave data. You can copy the parameters and data of the original WaveSample, or just the parameters. When copying within an instrument, selecting PARAMS ONLY helps conserve memory.

- Select COPY WAVESAMPLE. Press ***Enter•Yes***.
- The display reads TO INST=UNNAMED. Press an ***Instrument•Sequence Track*** button to choose the destination instrument (if necessary). Press ***Enter•Yes***.
- The display reads TO LAYER=##. Using the ***Up/Down Arrow*** buttons, select a destination layer. Press ***Enter•Yes***.
- Use the ***Up/Down Arrow*** buttons to select COPY=PARAMS ONLY or COPY=PARAMS & DATA. Press ***Enter•Yes***. The display reads WAVESAMPLE ## CREATED.

CMD WAVE	DELETE WAVESAMPLE
	Press Command / Wave / 2

Use DELETE WAVESAMPLE to erase unneeded WaveSamples from an instrument in memory. For example, you can delete the drum sound WaveSamples that you don't need from a drum kit. Doing this can help you conserve memory.

- Be sure the WaveSample you want to delete is selected on the Edit page.
- Select DELETE WAVESAMPLE. Press ***Enter•Yes***.
- The display will read DELETE WAVESAMPLE ##? If this is the WaveSample you want to delete, press ***Enter•Yes***. If not, press ***Cancel•No***.

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

CMD WAVE	WAVESAMPLE INFORMATION
	Press Command / Wave / 3

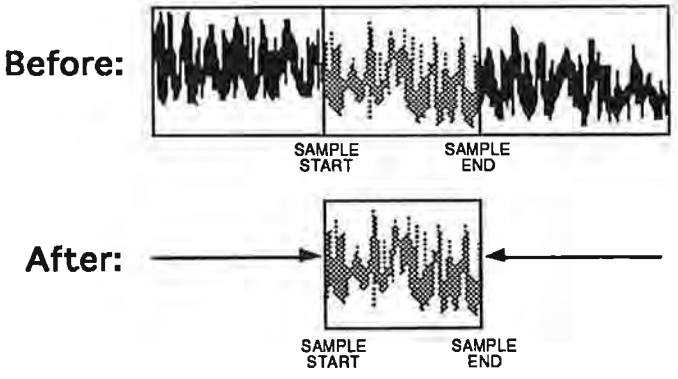
You can use WAVESAMPLE INFORMATION to name a WaveSample; find out the sample rate; whether or not a WaveSample is a copy; and what the root key is. The WAVESAMPLE INFORMATION command is used primarily for reference, and is useful for keeping track of things when you are developing sounds.

- Select WAVESAMPLE INFORMATION. Press ***Enter•Yes***.
- The display reads WS NAME=#####. Use the data entry controls to name the WaveSample as you would any other file. Press ***Enter•Yes***.
- The display reads SAMPLE RATE=##KHZ. You can now use the ***Up/Down Arrow*** buttons to change the sample rate at which the sample will play. Notice that changing the sample rate effectively changes the playback pitch as well. Press ***Enter•Yes***.
- The display reads ROOT KEY=##. The ***Up/Down Arrow*** buttons will now change the root key. Press ***Enter•Yes***.
- If the WaveSample is a copy, the display will read COPY OF LYR ## WS ##. This parameter is display only, and cannot be changed.

If you get the message, LAYER NOT IN PATCH, press ***Edit***, then ***Instrument***. Use the ***Arrow*** buttons to enable the layer in the current patch.

CMD WAVE	TRUNCATE WAVESAMPLE
	Press Command / Wave / 4

Truncating removes data before the sample start, and/or after the sample end at the same time. The memory formerly used by the truncated data is then returned to the available memory.



- Select TRUNCATE WAVESAMPLE. Press ***Enter•Yes***.

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

About Loop Commands

In the following loop commands, the messages INCREMENT LOOPSTART, or DECREMENT LOOPEND may appear. When they do, the ASR–10 is letting you know there's a problem with the loop length. Try changing the FADE ZONE. You could also press ***Edit***, then ***Wave***, and adjust the loop points. To move the loop points one sample at a time, make sure AUTO-LOOP FINDING (on the Edit/System•MIDI page) is OFF.

Note: Whenever the KEEP=OLD NEW message appears, the ASR–10 is giving you the chance to "audition" the results of the operation, before you commit any changes to memory.

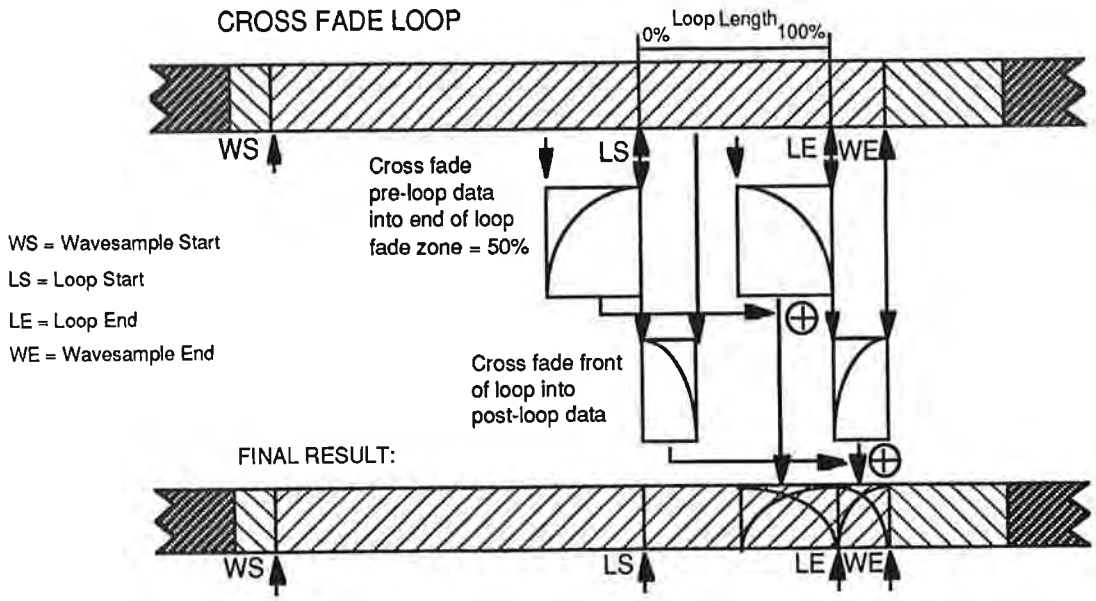
CMD WAVE	**CROSS FADE LOOP**
	Press Command / Wave / 5

The idea behind cross fading is to blend the data preceding the start of the loop with the data at the end of a loop, for a smooth transition between the two. The CROSS FADE LOOP command is a simple cross fade that works well with periodic waveforms that are relatively constant in tone. The default values are: SCALE DEPTH=3 dB; FADE ZONE=50%.

SCALE DEPTH determines the amount of volume drop that will have occurred in the original material at the halfway point of the cross fade. The material that is being added will follow an inverse curve, moving up to the same level at the halfway point. The result should be an even level of volume as the cross fade occurs.

The FADE ZONE determines the percentage of the loop over which the cross fade will occur. A setting of 99% would cause the cross fade to begin at the beginning of the loop. Lower values will start the fade nearer the *end* of the loop, causing the fade to occur in a shorter period of time.

Manual Ranges: FADE ZONE=1% to 99%; SCALE DEPTH=3 dB to 6 dB. With manual ranges, 3 dB will give you an exponential curve, while 6 dB is linear.

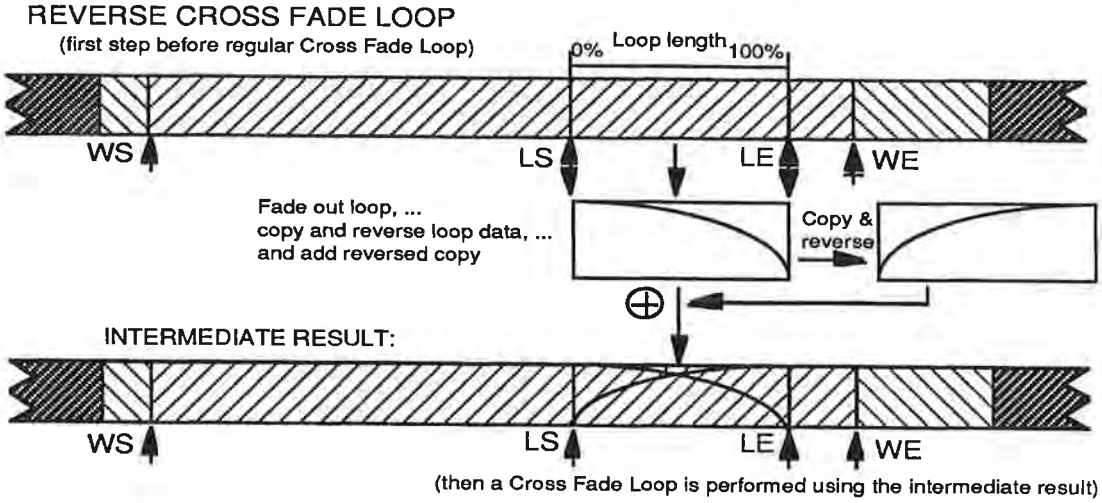


- Select CROSS FADE LOOP. Press ***Enter•Yes***.
- The display reads USE DEFAULT VALUES? Press ***Enter•Yes***.
- The display reads KEEP=OLD/NEW. Underline NEW and play the note to hear the new loop, or underline OLD to hear the original. Underline NEW and press ***Enter•Yes*** to keep the new loop. Or, press ***Cancel•No*** to keep the old loop (the old loop will be kept no matter the location of the cursor).
- Pressing ***Cancel•No*** in response to USE DEFAULT VALUES, prompts the ASR–10 to display the manual values for this command. Use the ***Left/Right Arrow*** buttons to scroll to select: SCALE DEPTH=(3.0, 3.5, 4.0, 4.5, 5.5, or 6.0 DB) and FADE ZONE (PERCENT)=(1 to 99). Use the ***Up/Down Arrow*** buttons, or the ***Data Entry Slider*** to change the scale depth or fade zone. Press ***Enter•Yes***.
- The display reads KEEP= OLD/NEW. Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to keep the original).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

CMD WAVE	REVERSE CROSS FADE
	Press Command / Wave / 6

The REVERSE CROSS FADE loop command helps eliminate tonal bumps, and smoothes out the tonal content of the loop. REVERSE CROSS FADE works well with multiple source sounds such as string sections, brass ensembles, and choirs. The default and manual values are the same as in the CROSS FADE LOOP command.



- Select REVERSE CROSS FADE. Press ***Enter•Yes***.
- The screen will ask, USE DEFAULT VALUES? Press ***Enter•Yes***.
- Pressing ***Cancel•No*** in response to USE DEFAULT VALUES, prompts the ASR–10 to display the manual values for this command. Use the ***Left/Right Arrow*** buttons to scroll to select SCALE DEPTH=## DB and FADE ZONE (PERCENT)=##. Press ***Enter•Yes***.
- The display reads KEEP=OLD NEW. Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to keep the original).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

CMD WAVE	ENSEMBLE CROSS FADE
	Press Command / Wave / 7

ENSEMBLE CROSS FADE is essentially a cross fade loop with the FADE ZONE set to 100%. If you use the default values, the ASR–10 may tell you DEFAULTS CANNOT BE SET. Then it will advise, INCREMENT LOOP START or, DECREMENT LOOP END (Edit/Wave page). This is because the data preceding the loop is shorter than the loop itself. For ENSEMBLE CROSS FADE to work, the fade zone must be 100% of the loop length. You can either make sample start smaller so that you add more data preceding the loop, or move the loop around using loop position on the Edit/Wave page. The only manual value here is SCALE DEPTH which is the same as in the CROSS FADE LOOP command. The fade zone is not a parameter, because it is always assumed to be 100% of the loop length.

- Select ENSEMBLE CROSS FADE. Press ***Enter•Yes***.
- The display reads USE DEFAULT VALUES? Press ***Enter•Yes***.
- Pressing ***Cancel•No*** in response to USE DEFAULT VALUES, prompts the ASR–10 to display the manual values for this command. Use the ***Left/Right Arrow*** buttons to scroll to select SCALE DEPTH=## DB. Press ***Enter•Yes***.
- The display reads KEEP=OLD/NEW. Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to keep the original).

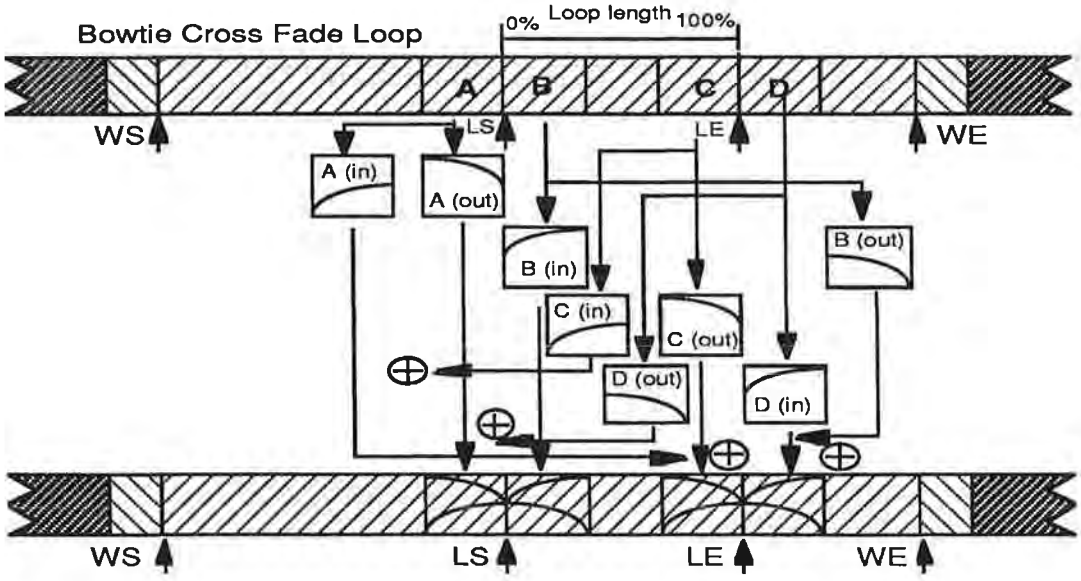
Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

CMD WAVE	BOWTIE CROSS FADE LOOP
	Press Command / Wave / 8

The BOWTIE CROSS FADE LOOP command is used to equalize the data around the loop points so the transition in to and out of the loop splice point is very smooth. The BOWTIE CROSS FADE LOOP works well on sounds with a lot of ambience, reverb, or other effects. The FADE ZONE can never be greater than 50%, otherwise you will overlap data halfway through the loop. The BOWTIE CROSS FADE is limited by the length of the post-loop data (the data from loop end to WaveSample end).

If you truncate a sound with WaveSample end equal to loop end, you can not use the bowtie loop, because it needs some data after loop end to make the loop. There would not be enough post-loop data to follow the command. If this happens, the ASR–10 will advise, DECREMENT LOOPEND. For the bowtie you need to have the loop away from the ends of the WaveSample. You can decrement loop position in the same way you had incremented it for the ensemble cross fade.

The default values are the same as the cross fade. Manual value ranges are: FADE ZONE (PERCENT): 1% to 50%; SCALE DEPTH: 3.0 dB to 6.0 dB.



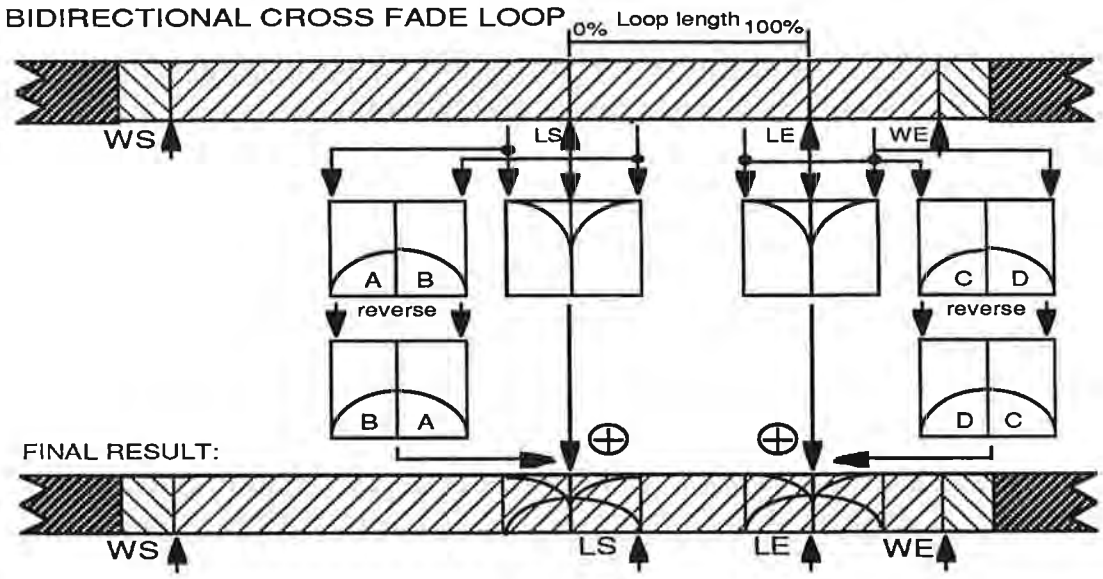
- Select BOWTIE CROSS FADE LOOP. Press ***Enter•Yes***.
- The display reads USE DEFAULT VALUES? Press ***Enter•Yes***.
- Pressing ***Cancel•No*** in response to USE DEFAULT VALUES, prompts the ASR–10 to display the manual values for this command. Use the ***Left/Right Arrow*** buttons to scroll to select: SCALE DEPTH=## DB, and FADE ZONE (PERCENT)=##. Press ***Enter•Yes***.
- The display reads KEEP=OLD/NEW. Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to keep the original).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

CMD WAVE	**BIDIRECTIONAL X-FADE**
	Press Command / Wave / 9

BIDIRECTIONAL X-FADE is optimized for use with bidirectional loops. One advantage of the bidirectional loop (which you can select by setting MODE= LOOP BIDIRECTION on the Edit/Wave page) is that it makes a loop that sounds twice as long as the amount of data being used. It does this by playing the loop forward and backward. As a result, the bidirectional loop is very memory conservative and is best suited for ensemble sounds that have a slowly modulated sustain, such as the bowing in a string section.

The default and manual values are the same as the cross fade. It is very important to remember that the loop points should be adjusted with autolooping off. Unlike other looping algorithms that look for a "zero crossing," BIDIRECTIONAL X-FADE works best when the loop points are set to the peaks in the waveform.



- Select BIDIRECTIONAL X-FADE. Press ***Enter•Yes***.
- The display reads USE DEFAULT VALUES? Press ***Enter•Yes***.
- Pressing ***Cancel•No*** in response to USE DEFAULT VALUES, prompts the ASR–10 to display the manual values for this command. Use the ***Left/Right Arrow*** buttons to scroll to select: SCALE DEPTH=## DB, and FADE ZONE (PERCENT)=##. Press ***Enter•Yes***.
- The display reads KEEP= OLD/NEW. Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to keep the original).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

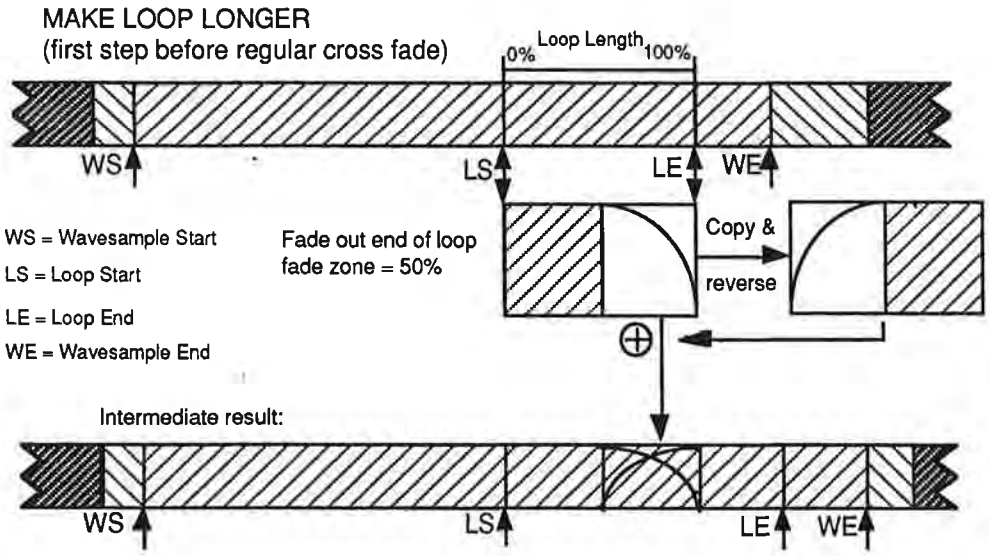
A Note About the COPIES WILL CHANGE Message

When using the following MAKE LOOP LONGER, SYNTHESIZE LOOP, or CONVERT SAMPLE RATE commands, you should select only the layer that contains the WaveSample you want to work on. If there are copies of the WaveSample you are editing elsewhere in the instrument, the ASR–10 will display the following message:

For instance, if you try to execute the CONVERT SAMPLE RATE command and the COPIES WILL CHANGE-OK? message appears, it is possible the conversion may produce apparent pitch problems while the Audition page is displayed. This is because the pitch parameters haven't been converted yet. However, they will be converted once you decide to choose KEEP=NEW. With the MAKE LOOP LONGER and SYNTHESIZE LOOP commands, the loops may sound odd until you choose KEEP=NEW.

CMD WAVE	**MAKE LOOP LONGER**
	Press Command / Wave / scroll using the arrow buttons

If you are having a hard time getting a satisfactory loop—after you've tried the other looping commands—try making the loop longer. This process physically enlarges the loop, making the periodicity of the loop less noticeable. However, there is a trade-off: longer loops use more memory. The default and manual values are the same as in the CROSS FADE LOOP command. *This is not recommended for bidirectional loops.*



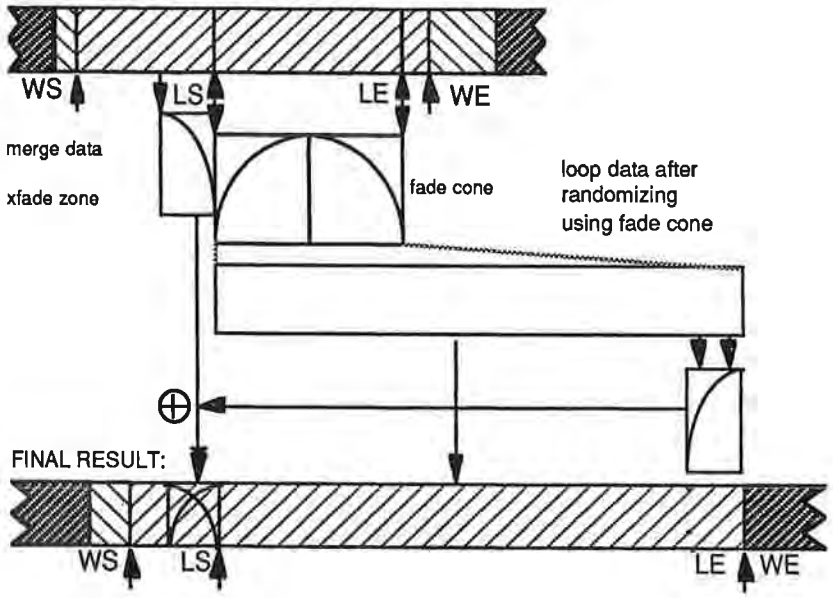
- Select MAKE LOOP LONGER. Press ***Enter•Yes***.
- The display reads USE DEFAULT VALUES? Press ***Enter•Yes***.
- Pressing ***Cancel•No*** in response to USE DEFAULT VALUES, prompts the ASR–10 to display the manual values for this command. Use the ***Left/Right Arrow*** buttons to scroll to select: SCALE DEPTH=## DB, and FADE ZONE (PERCENT)=##. Press ***Enter•Yes***.
- The display reads KEEP=OLD/NEW. Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to keep the original).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

CMD WAVE	SYNTHESIZED LOOP
	Press Command / Wave / scroll using the arrow buttons

This is the looping algorithm to use when all other looping techniques have failed to make a smooth loop. Unlike the other looping algorithms, the "synth loop" uses a certain amount of random data processing. It's a "loop anything" algorithm, which works well with special effects. With the SMOOTHNESS parameter, the smoother the loop, the more processes the ASR–10 goes through to create the synth loop. For example, if you do synth loop on a voice sample, and the SMOOTHNESS is set to very coarse, it may create a loop that sounds irregular or nervous. At the other extreme, if the setting is too fine, it may sound metallic. The default values are the same as in the CROSS FADE LOOP command. Manual values: FADE ZONE=1% to 99%. SMOOTHNESS=Very Fine to Very Coarse.

SYNTHESIZED LOOP



- Select SYNTHESIZED LOOP. Press ***Enter•Yes***.
- The display reads USE DEFAULT VALUES? Press ***Enter•Yes***.
- The display reads KEEP=OLD NEW. Press ***Enter•Yes***.
- Pressing ***Cancel•No*** in response to USE DEFAULT VALUES, prompts the ASR–10 to display the manual values for this command. Use the ***Left/Right Arrow*** buttons to scroll to select: FADE ZONE (PERCENT)=##, and SMOOTHNESS=VERY FINE, FINE, MEDIUM, COARSE, VERY COARSE. Press ***Enter•Yes***.
- The display reads KEEP=OLD NEW. Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to keep the original).

Hint: If you've done all you can to create a smooth sounding loop, and there are still irregular volume changes, try setting LOOP START and LOOP END (on the Edit/Wave page) to different values , and redoing this command.

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

CMD WAVE	CONVERT SAMPLE RATE
	Press Command / Wave / scroll using the arrow buttons

This command effectively resamples a WaveSample at a different sampling rate. You can start sampling with a high rate, then if you need more memory later, you can convert to a lower sample rate. The ASR–10 lets you compare these conversions before committing them to memory. When you convert from a higher to a lower sample rate, it's like transposing the WaveSample up the keyboard: you play fewer samples and the sound is shorter. Conversely, a lower to a higher sample rate has the effect of transposing down the keyboard: the sound gets bigger, and stretches out.

SAMPLE RATE Range: 48.0 kHZ to 6.25 kHZ.

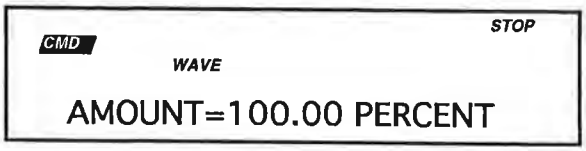
- Select CONVERT SAMPLE RATE. Press ***Enter•Yes***.
- The display reads SAMPLE RATE=## KHZ. Use the ***Data Entry Slider*** or ***Up/Down Arrow*** buttons to select a new sample rate. Press ***Enter•Yes***.
- The display reads KEEP=OLD NEW. Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to keep the original).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

CMD WAVE	TIME COMPRESS/EXPAND
	Press Command / Wave / scroll using the arrow buttons

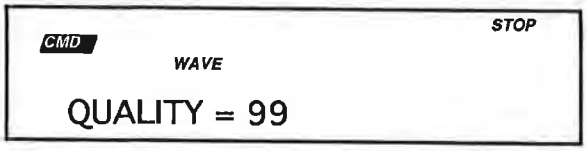
The TIME COMPRESS/EXPAND command shortens or lengthens the WaveSample duration, without affecting pitch. This is useful for "stetching" a WaveSample to fit a particular tempo, for example. To use this command:

- Press the ***Enter•Yes*** button after choosing a WaveSample to be edited. The display shows:



Range: 50 to 250
Values less than 100 shorten the duration, and values greater than 100 lengthen the duration.

- Press ***Enter•Yes*** , or scrolling right will display the QUALITY screen:



Range: 0 to 99
Higher values offer better fidelity on the compressed/expanded WaveSample, however, it takes longer to process this command.

Hint: Set the QUALITY parameter to 0 (or a low level) when checking the duration time for your particular application. This allows you to quickly verify that the time duration is correct, because it takes the least amount of time to process the data. When you know the exact duration amount, set the QUALITY parameter to a higher value, for improved fidelity. Bear in mind that the higher the QUALITY value, the longer it will take to process.

- Pressing ***Enter•Yes*** from the QUALITY screen will execute the Command. The display will read:

Range: 0 to 99
The xx value field counts up in real time. Pressing ***Cancel•No*** will abort the command.

If the selected WaveSample is a copy of another (or if there are copies of this WaveSample), the following screen will appear:

- Pressing ***Enter•Yes*** will take you to the PERCENT DONE screen, as explained earlier.

Note: After you have compressed/expanded your WaveSample, you may need to reset the Edit/Wave, LOOP START and LOOP END points (because the length of the WaveSample has changed).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

CMD WAVE	**COPY WAVE PARAMETERS**
	Press Command / Wave / scroll using the arrow buttons

This command will copy all the WaveSample parameters (but not the wave data) from a source WaveSample to a destination WaveSample. All the parameters (envelopes, filters, amp, pitch, mod parameters, etc.) are copied from the source WaveSample to the destination WaveSample, except for the actual sample pointers (sample start/end, loop start/end) and root key.

- Select COPY WAVE PARAMETERS. Press ***Enter•Yes***.
- The display reads TO INST=## L=## WS=##. Select the destination instrument, layer, and WaveSample. The current (or Edit) WaveSample is used as the source.
- Press ***Enter•Yes***. The display reads COMMAND COMPLETED.

Command/Amp Page (Amplitude Commands)

CMD AMP	NORMALIZE GAIN Press Command / Amp / 0

This command maximizes the dynamic range of a WaveSample. NORMALIZE GAIN always works on the currently selected WaveSample (selected on the Edit page) and has no parameters. It will make the sound louder, unless it is already at full level. NORMALIZE GAIN finds the largest sample value, and uses that to scale the whole range of the sample to its absolute full level.

Before: **After:**

Press ***Enter•Yes*** to invoke the command. The display will read DATA BEING PROCESSED, followed by COMMAND COMPLETED.

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

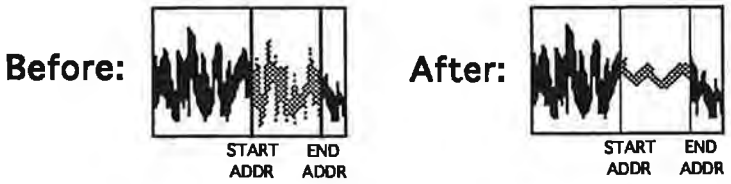
For more information, see **Managing Your System Headroom** in *Section 1 — Controls and Architecture.*

CMD AMP	VOLUME SMOOTHING
	Press Command / Amp / 1

VOLUME SMOOTHING works like a dynamic compressor. The idea is to remove amplitude changes. The degree to which it does this is determined by "smoothness." It works on a given range of sound, determined by the start and end address parameters. This command is very useful for smoothing out a loop, and getting rid of clicks caused by volume bumps.

For instance, if you are working with a piano loop, there is a natural envelope that is always decaying. If you try to loop on the section that is decaying, the loop end data will be at a much lower amplitude than the beginning. You would end up with a click, because of an abrupt change in amplitude. VOLUME SMOOTHING will help make the level constant. When the ASR–10 displays DATA RANGE TOO SMALL, it is advising you that the minimum range to perform this command is 256 samples.

This command is most useful for loops, but you can use it on the whole sample. If you wanted to remove a natural envelope from a piano, for example, you can use volume smoothing over the whole length of the sample. You can then superimpose any amplitude envelope from the ASR–10 over that sample. Value ranges: SCALE DEPTH=3.0 to 6.0 dB. SMOOTHNESS=VERY FINE to VERY COARSE.



- Select VOLUME SMOOTHING. Press ***Enter•Yes***.
- The display reads USE LOOP FOR RANGE? Press ***Enter•Yes*** to automatically set the start and end addresses to the loop points.
- Select START ADDR=## (##). Use the ***Left/Right Arrow*** buttons to scroll through the various screens. Use the ***Up/Down Arrow*** buttons to change the values in the display.
- Scroll to and select END ADDR=## (##). Scroll to and select the SCALE DEPTH=## DB. Scroll to and select the SMOOTHNESS=####. Press ***Enter•Yes***.
- Select KEEP=OLD NEW. Press ***Enter•Yes*** to complete the command.
- Pressing ***Cancel•No*** for USE LOOP FOR RANGE? lets you adjust the start and end addresses manually.
- Select START ADDR=## (##). Scroll to and select the END ADDR= ## (##).
- Scroll to and select the SCALE DEPTH=## DB. Scroll to and select the SMOOTHNESS=####. Press ***Enter•Yes***.
- The display reads KEEP=OLD NEW. Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to keep the original).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

About Mix, Merge, and Splice

The MIX, MERGE, and SPLICE commands are very similar. You will always be working on a pair of WaveSamples—source and destination WaveSamples. These commands take a source WS and mixes, merges, or splices it into a destination WS. FR INST=## L=## WS=##, determines the source. TO INST=## L=## WS=## determines the destination. For best results the source and destination WaveSample should be in the same instrument, in different layers.

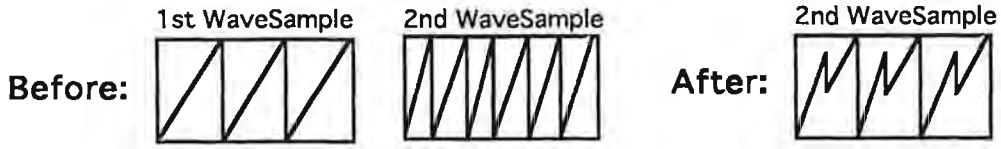
CONVERT SOURCE PITCH and CONVERT DEST PITCH let you combine WaveSamples that were sampled at different sample rates, or were on different root keys. If the WaveSamples are on the same root key, but have different sampling rates, their data will be different. The ASR–10 can't simply add them together because they would be out of tune with each other. It has to convert the sample rate of one or the other.

Before executing these commands, you should assign the layer containing the destination WaveSample to a different patch (on the Edit/Instrument page) than the patch containing the source. This way, when you "audition" the sound after the commands have been executed, you can specifically hear the final results of the operation, and not the final results and the source WaveSample together. This process is similar to that described in the COPY WAVESAMPLE command.

Note: When using the MERGE or SPLICE commands, the VOLUME BALANCE function only lets you hear the relative volumes of the source and destination WaveSamples, not the final results. You will be able to hear the final results after the WaveSamples have been merged or spliced. VOLUME BALANCE controls the relative volumes of the source and destination WaveSamples. When the value is negative, the source WaveSample is louder. And when the value is positive, the destination is louder.

CMD AMP	MIX WAVESAMPLES
	Press Command / Amp / 2

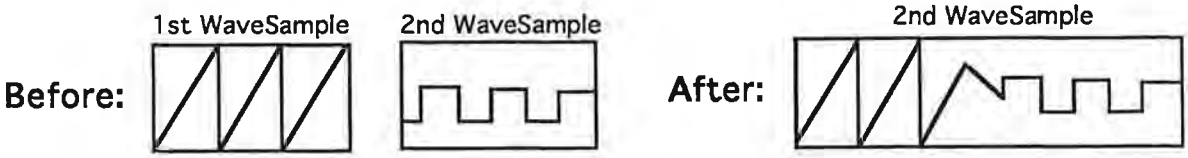
Mixing WaveSamples is a lot like layering two WaveSamples. It adds all of the wave data from the two WaveSamples together, resulting in a single composite WaveSample. For example, to make a "bowed voice," you would assign the "bow" WaveSample to one layer, then the "voice" WaveSample to a different layer, and use "mix" to combine them digitally into one WaveSample.



- Select MIX WAVESAMPLES. Press ***Enter•Yes***.
- The display reads ARE SAMPLES LAYERED? Press ***Enter•Yes***.
- Select FR INST=## L=## WS=##.
- Scroll to TO INST=## L=## WS=##.
- Scroll to CONVERT SOURCE PITCH, or CONVERT DEST. PITCH. Press ***Enter•Yes***.
- Scroll to VOLUME BALANCE=(-99 to +99). Use the ***Data Entry Slider*** to set the balance between the two.
- Press ***Enter•Yes***. The display shows KEEP=OLD/NEW. Press ***Enter•Yes*** to keep the mixed WaveSamples, or ***Cancel•No*** to keep the originals intact.

CMD AMP	MERGE WAVESAMPLES
	Press Command / Amp / 3

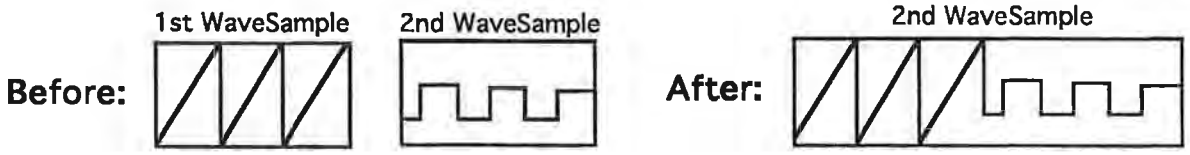
MERGE WAVESAMPLES takes source wavedata, and cross fades it into destination wavedata. MERGE WAVESAMPLES has many of the same parameters as MIX, with the addition of SCALE DEPTH and FADE ZONE. The SCALE DEPTH and FADE ZONE operate the same as with the cross fade command.



- Select MERGE WAVESAMPLES. Press ***Enter•Yes***.
- The display reads ARE SAMPLES LAYERED? Press ***Enter•Yes***.
- Select FR INST=## L=## WS=##. Scroll to TO INST=## L=## WS=##.
- Scroll to SCALE DEPTH=## DB. Scroll to FADE ZONE (PERCENT)=##.
- Scroll to CONVERT SOURCE PITCH or CONVERT DEST. PITCH.
- Scroll to VOLUME BALANCE=##. Press ***Enter•Yes***.
- The display reads KEEP=OLD/NEW. Press ***Enter•Yes*** to keep the merged WaveSamples, or ***Cancel•No*** to keep the originals intact.

CMD AMP	SPLICE WAVESAMPLES Press Command / Amp / 4

This command takes the end of a source sample, and splices it to the beginning or start of the destination sample. SPLICE WAVESAMPLES is just like a tape splice—a butt splice. You can also think of it as MERGE with a zero length FADE ZONE.



- Select SPLICE WAVESAMPLES. Press ***Enter•Yes***.
- The display will ask ARE SAMPLES LAYERED? Press ***Enter•Yes***.
- Select FR INST=## L=## WS=##. Scroll to TO INST=## L=## WS=##.
- Scroll to CONVERT SOURCE PITCH. Scroll to CONVERT DEST. PITCH.
- Scroll to VOLUME BALANCE= (-99 to +99). Use the ***Data Entry Slider*** to set the balance between the two.
- Press ***Enter•Yes***. The display reads KEEP=OLD/NEW. Press ***Enter•Yes*** to keep the spliced WaveSamples, or ***Cancel•No*** to keep the originals intact.

CMD AMP	FADE IN
	Press Command / Amp / 5

This command scales the volume of the wave data from 0% to 100% to create a digital FADE IN of the sound. First, select a data range, using the start and end addresses, then choose an appropriate fade curve. 3 dB fades are very smooth and analogous to the way our hearing responds to sound.

- Select FADE IN. Press ***Enter•Yes***.
- Select START ADDR=## (##). Scroll to END ADDR=## (##).
- Scroll to SCALE DEPTH=## DB. Press ***Enter•Yes***.
- The display reads KEEP=OLD NEW.
- Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to keep the original).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

CMD AMP	FADE OUT
	Press Command / Amp / 6

This command scales the volume of the wave data from 100% to 0% to create a digital FADE OUT of the sound. The parameters are the same as above.

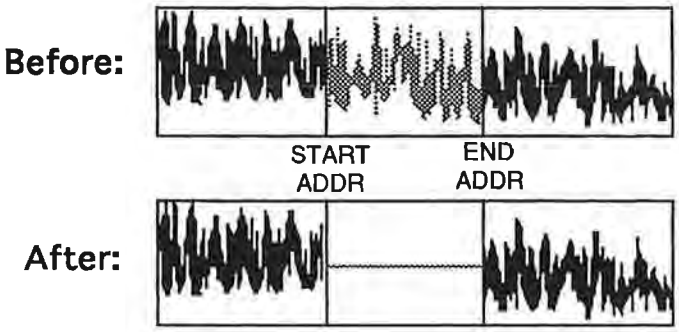
- Select FADE OUT. Press ***Enter•Yes***.
- Select START ADDR=## (##). Scroll to END ADDR=## (##).
- Scroll to SCALE DEPTH=## (##). Press ***Enter•Yes***.
- The display reads KEEP=OLD NEW.
- Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to keep the original).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

Command/LFO Page (Data Commands)

CMD LFO	CLEAR DATA
	Press Command / LFO / 0

Use this command when you want silence as part of the WaveSample. Use the "start and end" addresses to select the section of WaveSample data you want to clear.

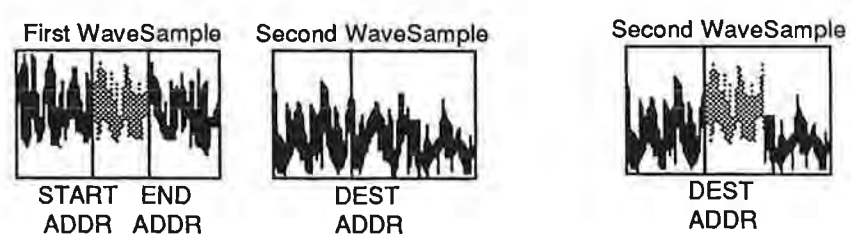


- Select CLEAR DATA. Press ***Enter•Yes***.
- Select START ADDR=## (##). Scroll to END ADDR=## (##).
- Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to abort the procedure).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

CMD LFO	COPY DATA
	Press Command / LFO / 1

COPY DATA enables you to copy a section of wave data from one WaveSample and paste it over part of another WaveSample, replacing the data that was previously there. Use the "start and end" addresses, to copy wave data from a source to a destination. In fact, source and destination could be the same WaveSample, or the destination WaveSample could be in a different instrument.

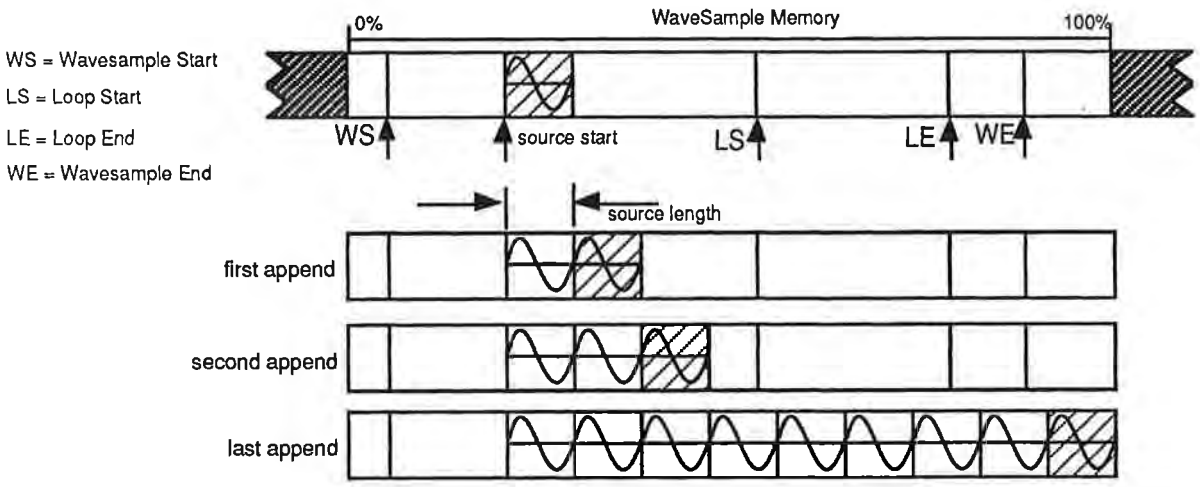


- Select COPY DATA. Press ***Enter•Yes***.
- Select FR INST=## L=## WS=##. Scroll to START ADDR=## (##).
- Scroll to END ADDR=## (##). Scroll to TO INST=## L=## WS=##.
- Scroll to DEST ADDR=## (##). Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to abort the procedure).

CMD LFO	**REPLICATE DATA**
	Press Command / LFO / 0

The REPLICATE DATA command is used to append a section of wave data to itself as many times as will fit within the original WaveSample size. The replicated data will fill the remaining area of the WaveSample that lies after the END ADDR. Set the data range with the "start and end" addresses.

REPLICATE DATA

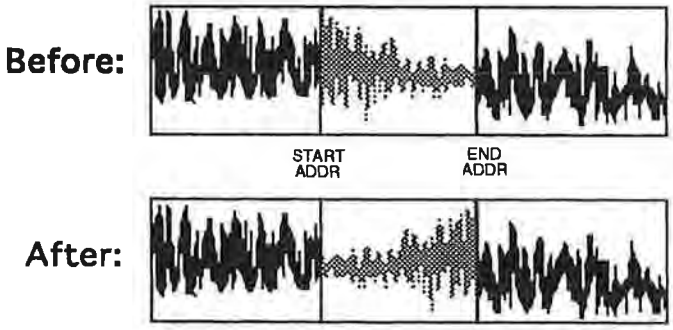


- Select REPLICATE DATA. Press ***Enter•Yes***.
- Select START ADDR=## (##). Scroll to END ADDR=## (##). Press ***Enter•Yes***.
- The display reads KEEP=OLD NEW. Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to abort the procedure).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

CMD LFO	REVERSE DATA
	Press Command / LFO / 3

The REVERSE DATA command takes the wave data in the specified range and turns it around backward, so that the last sample becomes the first one, the next-to-last sample becomes the second, and so on. It has the same effect as playing the WaveSample backward.

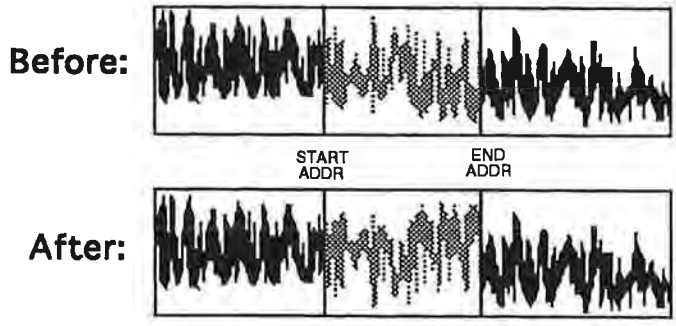


- Select REVERSE DATA. Press ***Enter•Yes***.
- Select START ADDR=## (##). Scroll to END ADDR=## (##).
- Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to abort the procedure).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

CMD LFO	INVERT DATA
	Press Command / LFO / 4

This command inverts the phase of the selected wave data. This is the equivalent of playing the WaveSample upside down, with all frequency peaks becoming valleys and vice versa. If you're performing advanced WaveSample manipulation it is sometimes helpful to invert the phase of a signal for the purposes of cross fading. This will sometimes produce a smoother cross fade point.

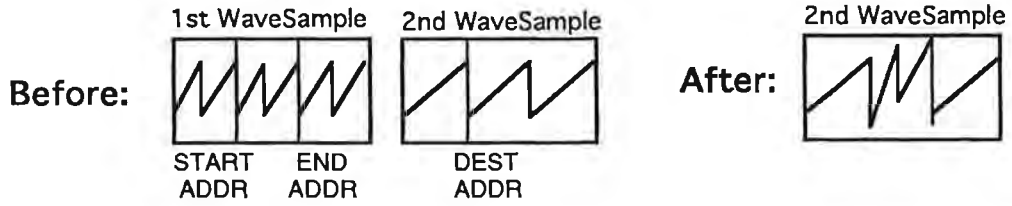


- Select INVERT DATA. Press ***Enter•Yes***.
- Select START ADDR=## (##). Scroll to END ADDR=## (##).
- Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to abort the procedure).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

CMD LFO	**ADD DATA**
	Press Command / LFO / 5

This command is similar to COPY DATA, except it adds a section of the wave data from one WaveSample into part of another WaveSample, creating a composite of the two, according to the source and destination ranges. It could be the same WaveSample, or two different ones. When PREVENT CLIPPING is enabled, the ASR–10 (if necessary) will scale down the destination and source data, so the added data doesn't result in clipping.



- Select ADD DATA. Press ***Enter•Yes***.
- Select FR INST=## L=## WS=##. Scroll to START ADDR=## (##).
- Scroll to END ADDR=## (##). Scroll to TO INST=## L=## WS=##.
- Scroll to DEST ADDR=##(##). Scroll to PREVENT CLIPPING=(NO/YES). Press ***Enter•Yes***.
- The display reads KEEP=OLD NEW. Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to abort the procedure).

CMD LFO	**SCALE DATA**
	Press Command / LFO / 6

This command is a "sound sculpting" tool that will enable you to shape the volume contour of the waveform. As in the other commands, you select some range of data using the start and end addresses. You also set the scaling range with SCALE START and SCALE END points. These are multipliers. SCALE START is a multiplier used at the beginning of the source range. SCALE END is the multiplier used at the end of the source range. If the two scale points are the same, it's like multiplying the whole range of data by the same number. So, if you wanted to drop the level in half, you would set SCALE START and END to 0.5. SCALE DEPTH works as described previously. If the scale points are different, the data will be scaled using a scaling curve whose depth is controlled by the SCALE DEPTH.

- Select SCALE DATA. Press ***Enter•Yes***.
- Select START ADDR=## (##). Scroll to END ADDR=## (##). Scroll to SCALE START=##.
- Scroll to SCALE END=##. Scroll to SCALE DEPTH=## DB. Press ***Enter•Yes***.
- The display reads KEEP=OLD NEW. Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to abort the procedure).

Note: When STEREO LAYER LINK= ON, this command will be simultaneously performed on the corresponding WaveSample in the companion layer.

WaveSample & Layer Parameters

Section 11 — WaveSample and Layer Parameters

This section covers the parameters which can be edited independently for each individual WaveSample within an Instrument. For a general overview of the concepts involved, refer to *Section 9 — WaveSample and Layer Concepts*.

Edit/Pitch Page

Use parameters on this page to adjust the pitch of the selected WaveSample(s).

Note: When STEREO LAYER LINK= ON, all pitch modulation in the right layer will be slaved to the settings in the WaveSamples assigned to the same or overlapping key zones in the left layer. This includes BEND RANGE, NOISE RATE, and all pitch modulation. Pitch modulation settings for WaveSamples in the right layer will be ignored.

EDIT PITCH	**ROOT KEY - FINE (Fine Tune)**
	Press Edit / Pitch / 1

These parameters allow you to change the WaveSample root key and adjust its fine tuning. The root key is the key at which the WaveSample plays back at the same pitch as it was recorded. Scroll right to select the fine tune range.

Root Key Range: KY0 to K11, C0 to G9
Fine Tune Range: -99 to +99 cents

Note: Keys in the octave below C0 are displayed as KY0 through K11. These values are only accessible when the Instrument is transposed up.

EDIT PITCH	**LFO AMOUNT**
	Press Edit / Pitch / 2

This parameter allows you to set the amount by which the LFO will modulate pitch (for vibrato, etc.).

Range: -15.7 to +15.7 semitones

EDIT PITCH	**ENV 1 AMOUNT**
	Press Edit / Pitch / 3

This parameter allows you to set the amount by which ENV 1 will modulate pitch.

Range: -15.7 to +15.7 semitones

EDIT PITCH	NOISE RATE - AMOUNT
	Press Edit / Pitch / 5

These parameters control the NOISE generator. The noise generator produces a randomly changing level. Its main use is to randomize or "humanize" the pitch. You can also use it to modulate the filters, amplitude, etc. NOISE RATE determines how frequently the random levels will change. Use the ***Right Arrow*** button to select AMOUNT. Noise AMOUNT determines the amount by which the NOISE generator will modulate pitch.

Rate range: 0 to 99
Amount: -99 to +99

EDIT PITCH	BEND RANGE
	Press Edit / Pitch / 6

This parameter allows you to set the pitch bend range on the WaveSample level.

Range: 0 through 12 semitones, GLOBAL

Note: When BEND RANGE=GLOBAL the WaveSample will use the GLOBAL BEND RANGE setting on the Edit/System•MIDI page.

EDIT PITCH	PITCH MOD - AMOUNT
	Press Edit / Pitch / 7

Use this parameter to assign one of fifteen modulators to modulate pitch, and to set the amount (depth) of that modulator.

Range: -99 to +99

EDIT PITCH	WS RANGE LO - HI
	Press Edit / Pitch / scroll using the arrow buttons

This parameter determines the key range the selected WaveSample will cover. Select the WS RNG LO parameter, play the key on your MIDI controller that you want to be the low key, the cursor will then automatically switch to the HI parameter, which can then be set in the same fashion.

Range: KY0 to K11, C0 to G9

Note: Keys in the octave below C0 are displayed as KY0 through K11. These values are only accessible when the Instrument is transposed up.

Note: As with the instrument key range, if you accidentally set the High Key *below* the Low Key, the selected WaveSample will not be heard.

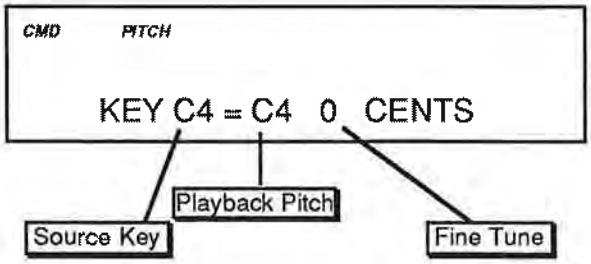
Command/Pitch Page

These commands allow you to create, delete and manipulate pitch tables. The ASR–10 can store up to eight user-defined pitch tables in each instrument. Each layer can use a different pitch table. Which pitch table a given layer will use is selected on the Edit/Layer page.

CMD PITCH	**EDIT PITCH TABLE**
	Press Command / Pitch / 0

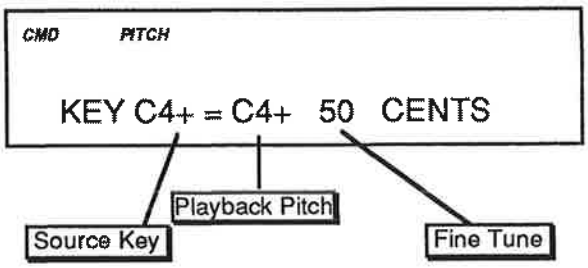
Use this command to create or edit a pitch table.

- Sample a sound or load a sound from disk. If you sampled a sound, pick C4 as the root key.
- Select EDIT PITCH TABLE.
- Press ***Enter•Yes***. When no pitch tables exist in the currently selected instrument, the display reads CREATE NEW PITCH TBL? Pressing ***Enter•Yes*** will assign the newly created pitch table to the currently selected layer. Note that when STEREO LAYER LINK= ON, this command will also assign the newly created pitch table to the companion layer.
- If multiple pitch tables exist in the current instrument, you must first assign a layer in the instrument to use the pitch table that you want to edit (with the PITCH TBL parameter on the Edit/Layer page), and then invoke the pitch table command. If you're editing an existing pitch table, the CREATE NEW PITCH TBL prompt will not appear.
- Press ***Enter•Yes***. The display reads KEY E4 = * 0 CENTS. This is the default display. If you press any key on the MIDI keyboard, the display will change. Pressing ***Cancel•No*** will abort this command.
- Press Middle C on the MIDI keyboard. The display now looks like this:



The source key can be thought of as the physical location of the key on the MIDI keyboard. The playback pitch can be the same as the source key—as shown in the display—or it can be any pitch from A0 to C8. The range of the fine tune is 0 to 99 cents. This allows you to create micro-tunings. Here's how:

- Play C4+ on the MIDI keyboard (the source key), or use the ***Up/Down Arrow*** buttons or the ***Data Entry Slider*** to select the source key.
- Press the ***Right Arrow*** button twice to move the cursor under the Fine Tune value, and change it to 50 CENTS.



You have just defined a new interval between C and C4+. The interval has now been stretched by 50 cents. You can now repeat this procedure for each key on the MIDI keyboard to create your own custom pitch table.

Press the ***Right Arrow*** button until the display shows TBL NAME= PITCHTABLE 1 (the default pitch table name). You can name your custom pitch table using the ***Left/Right Arrow*** buttons to move the cursor, and the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to select the characters. After you've named your custom pitch table:

- Press ***Enter•Yes*** to exit EDIT PITCH TABLE.

Note: For the following commands, you can't COPY, DELETE, or EXTRAPOLATE the STANDARD or NO PITCH pitch tables. If you try, the NO EDIT PITCH TABLE message will be displayed. You must create a new pitch table first using the EDIT PITCH TABLE command.

CMD PITCH	COPY PITCH TABLE
	Press Command / Pitch / 1

This command will copy a pitch table either within an instrument or from one instrument to another instrument. If you want to have multiple pitch tables within an instrument, start by selecting a layer that uses the pitch table to be copied. Use the COPY PITCH TABLE command to make a copy of the pitch table used by the currently selected layer. Next, assign a layer to use the pitch table copy (with the PITCH TBL parameter on the Edit/Layer page). Finally, with that layer selected, use the EDIT PITCH TABLE command to edit the pitch table copy to suit your needs. Remember to rename your newly edited pitch table to avoid confusion.

- Select COPY PITCH TABLE.
- Press ***Enter•Yes***. The display reads TO INST= (name). Select the instrument where you would like the pitch table to be placed. Each instrument can contain up to 8 different pitch tables.
- Press ***Enter•Yes***. The display reads PITCH TABLE (number) CREATED.

CMD PITCH	DELETE PITCH TABLE
	Press Command / Pitch / 2

This command will erase (delete) a pitch table. Any layers that used the pitch table that was deleted will have their Edit/Layer PITCH TBL parameter automatically set to use the (STANDARD) western equal temperment tuning.

CMD PITCH	EXTRAPOLATE PITCH TBL
	Press Command / Pitch / 3

Use this command to extend pitch table relationships across the keyboard. In other words, you can take the intervals defined between any two keys or group of keys, and extend that same relationship over the entire keyboard.

- Select EXTRAPOLATE PITCH TBL.
- Press ***Enter•Yes***. The display reads SOURCE LO=## HI=## (## is a key number). This determines the range of the extrapolation.
- Play the low key of the range you wish to extrapolate.
- Scroll right to underline HI=##.
- Play the high key of the range you wish to extrapolate.
- Press ***Enter•Yes***.

ASR–10 Filters

Each ASR–10 voice has its own pair of digital filters, F1 and F2, that are connected in series. The filter settings determine which frequencies will be allowed to pass through to the output.

Low-Pass and High-Pass

A low-pass filter allows only those frequencies below the filter cutoff frequency to pass—higher frequencies are filtered out. The reverse is true for the high-pass filters. In the ASR–10, the first filter (F1) is always a low-pass filter. The second filter (F2) can be either a high-pass or low-pass filter.

Poles — Rolloff Curves

"Pole" is an engineering term that describes the steepness of a filter. Each additional pole gives a filter a steeper rolloff curve. In the ASR–10, four poles are available, but they are divided between two filters; either as 2 and 2, or as 3 and 1. A single-pole filter will rolloff at 6 dB per octave; a 2-pole, 12 dB per octave; a 3-pole, 18 dB per octave; and 4-pole, 24 dB per octave. To reproduce a 4-pole low-pass filter (for that "classic" analog synth sound) you would set both filters to 2-pole low-pass. Each filter would then roll-off at 12 dB per octave, resulting in a total 24 dB per octave rolloff.

Cutoff Frequency

Every filter has a "knee" in the response curve, known as the cutoff frequency. This is the frequency at which the filter begins filtering out frequencies. The filter cutoff frequency can remain fixed over time, or it can be varied by modulating the filter with an envelope, LFO, velocity, etc. You can create some very interesting filter configurations by using a different modulator for each filter. For instance, try using pressure to modulate the filter. You can drive one filter up with pressure, while simultaneously driving the other down.

Edit/Filters Page

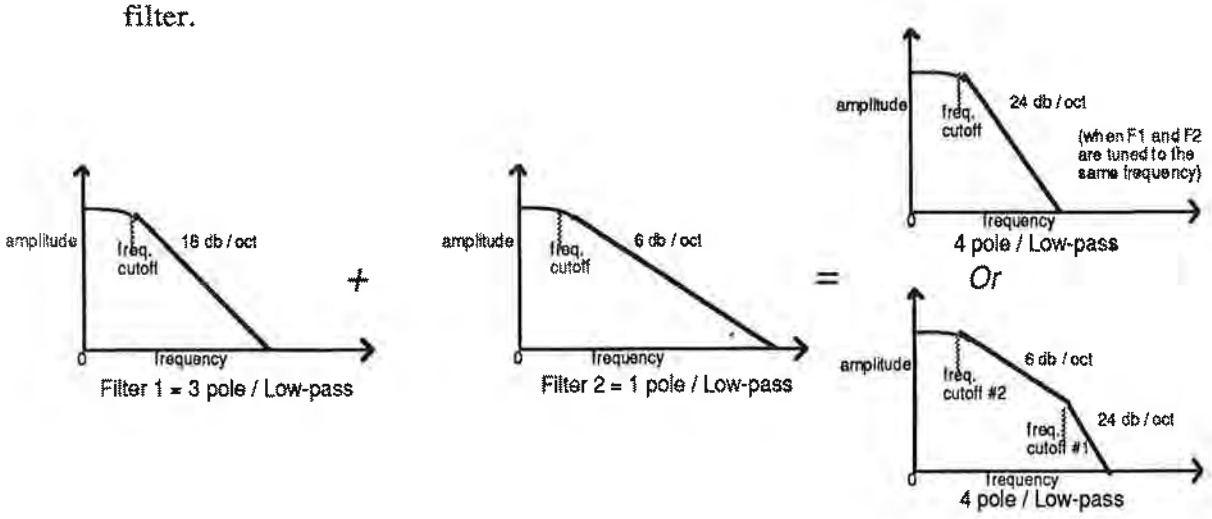
Use the parameters on this page to adjust the filters.

EDIT FILTERS	**MODE (Filter Mode)**
	Press Edit / Filters / 0

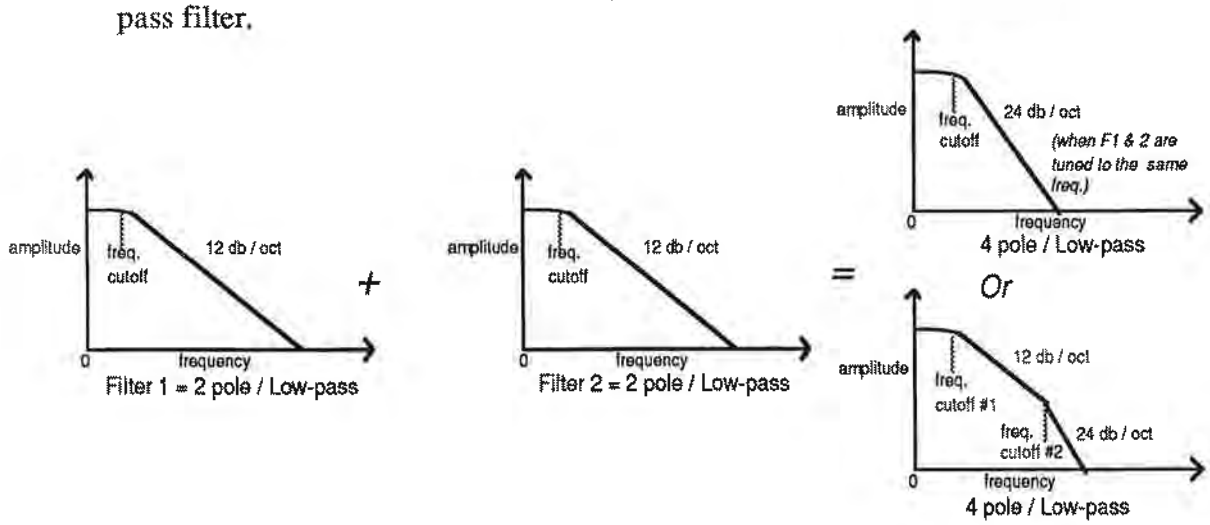
Use this parameter to choose one of the four filter modes.

The following diagrams show some possible filter configurations. On the left are the response curves of the two filters shown separately. On the right are some possible shapes of the combined filters.

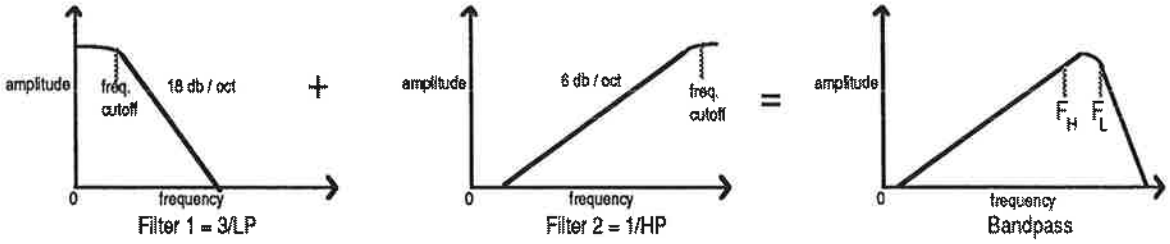
- MODE F1=3/LP F2=1/LP. F1 is a 3 pole low-pass filter. F2 is a 1-pole low-pass filter.



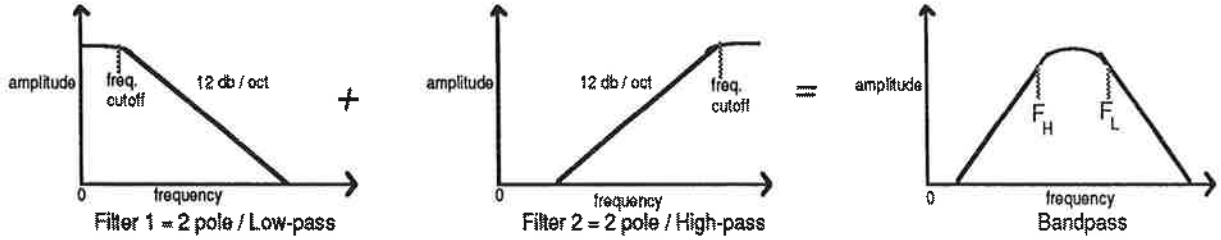
- MODE F1=2/LP F2=2/LP. F1 is a 2-pole low-pass filter. F2 is also a 2-pole low-pass filter.



- MODE F1=3/LP F2=1/HP. F1 is a 3-pole low-pass filter. F2 is a 1-pole high-pass filter.



- MODE F1=2/LP F2=2/HP. F1 is a 2-pole low-pass filter. F2 is a 2-pole high-pass filter.



EDIT FILTERS	**CUTOFF F1 (Filter 1) - F2 (Filter 2)**
	Press Edit / Filters / 1

This parameter allows you to set the initial filter cutoff frequency for each of the two filters. With a low-pass filter, a setting of 150 lets all the original signal pass through the filter. With a high-pass filter, a setting of 0 lets through all of the original signal. This setting represents the "filter floor," or manual level. The effect of any selected modulators will be added to (or subtracted from) this level.

Ranges: 0 to 150

EDIT FILTERS	**ENV2 AMT F1 (Filter 1) - F2 (Filter 2)**
	Press Edit / Filters / 2

Use this parameter to set the depth or amount by which Env 2 will modulate the filter cutoff frequency. Modulation amounts can be positive or negative.

Ranges: -99 to +99

EDIT FILTERS	**KBD AMOUNT F1 (Filter 1) - F2 (Filter 2)**
	Press Edit / Filters / 3

This parameter sets the amount by which the location of a note on the keyboard will modulate the filter cutoff frequency (filter keyboard tracking). This allows the relative brightness of a sound to remain constant over the keyboard. To have a filter track the keyboard exactly you should set KBD AMT to +50.

Ranges: -99 to +99

EDIT FILTERS	**F1 MOD - AMOUNT**
	Press Edit / Filters / 7

Use this parameter to select a modulation source and the modulation amount for the Filter 1 cutoff frequency. Any of the fifteen available modulators can be applied to modulate the Filter 1 cutoff frequency. Modulation amounts can be positive or negative.

Range: -99 to +99

EDIT FILTERS	**F2 MOD - AMOUNT**
	Press Edit / Filters / 8

This parameter serves the same function as F1 MOD - AMOUNT, but controls Filter 2. It operates in an identical fashion. Any of the fifteen available modulators can be applied to modulate the Filter 2 cutoff frequency.

Edit/Amp Page

The parameters on this page control the amplitude (or volume) of the WaveSample.

EDIT AMP	WS VOLUME - PAN
	Press Edit / Amp / 1 (for Volume) 2 (for Pan)

WaveSample Volume

Use this parameter to set the volume of an individual WaveSample. This parameter also controls the amount by which ENV 3 will modulate the WaveSample volume.

Range: 0 to 99

WaveSample Pan

Use this parameter to assign a WaveSample to a location in a stereo field, either within the main outputs, or any of the pairs of outputs on the optional Output Expander.

Range: -99 (full left) to +99 (full right)

EDIT AMP	VOLUME MOD - AMOUNT
	Press Edit / Amp / 7

This parameter allows you to assign a modulator to control the volume of an instrument, layer, and/or WaveSample. Any of the fifteen available modulators can be applied to modulate volume. Raising this value will lower the unmodulated volume of the WaveSample(s), creating the headroom necessary for the volume modulation. Note that the AMOUNT value is positive-going only. To obtain negative volume modulation, you must edit the A-B FADE IN and C-D FADE OUT values (see later in this Section).

Amount Range: 0 to 99

EDIT AMP	PAN MOD - AMOUNT
	Press Edit / Amp / 8

This parameter allows you to assign a modulator to control the panning of an instrument, layer, and/or WaveSample. Any of the fifteen available modulators can be applied to modulate pan.

Range: -99 to +99

EDIT AMP	A - B FADE IN - TO
	Press Edit / Amp / 3

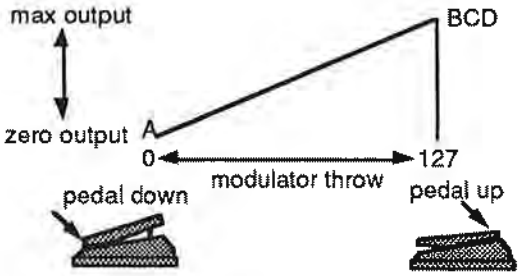
EDIT AMP	C - D FADEOUT - TO
	Press Edit / Amp / 4

Volume Modulator Cross Fade Breakpoints

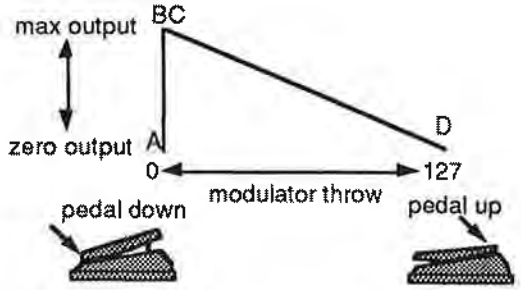
The cross fade breakpoints (A and B, C and D) allow you to scale the effect of the volume modulator (selected with the Edit/Amp VOLUME MOD parameter) to create keyboard, velocity, or other types of cross fades. Using the cross fade breakpoints, you can customize how the volume modulator will affect the volume of the WaveSample across the range (or throw) of its output. This lets you use velocity, keyboard position, pressure or any other modulator to cross fade between WaveSamples.

Think of the A-B FADE IN/C-D FADEOUT parameters as a trapezoid, whose shape will affect the output of the modulator assigned to the Edit/Amp VOLUME MOD parameter. The A-B FADE **IN** value sets the point along the throw of the volume modulator that the modulator's output value will begin to rise above zero (i.e. fade in). The A-B FADE **TO** value sets the point along the throw of the volume modulator that the modulator's output value will have risen to its maximum output. The C-D FADE **OUT** value sets the point along the throw of the volume modulator that the modulator's output value will begin to fall from maximum (i.e. fade out). The C-D FADE **TO** value sets the point along the throw of the volume modulator that the modulator's output value will have fallen to zero.

The default values are "A-B FADE IN=0 TO 127" and "C-D FADE OUT=127 TO 127," imposing the following shape on the output of the volume modulator. In this example the optional CV Pedal is the volume modulator:



To reverse the polarity of the volume modulator, set the values as follows: "A-B FADE IN=0 TO 0" and "C-D FADE OUT=0 TO 127." The shape imposed on the output of the volume modulation is as follows; again, the CV Pedal is the volume modulator:



Note: Whenever you cross fade between WaveSamples, the WaveSamples must be in different layers, since only one WaveSample can occupy a given key within a single layer.

See *Section 16 — Instrument Programming Applications* for information on programming keyboard cross fades.

EDIT AMP	**FADECURVE**
	Press Edit / Amp / 5

This parameter determines what type of curve will be used when fading samples in and out between breakpoints A-B and C-D. There are two possible fadecurves:

- CROSSFADE — This convex curve features a steep initial rise/drop off. It is best used when a quick change between breakpoints is desired, as in a keyboard or velocity cross fade.
- LINEAR — This linear curve ramps up and down evenly, making it preferable when smoother transitions between breakpoints are appropriate, such as when using pressure to cross fade.

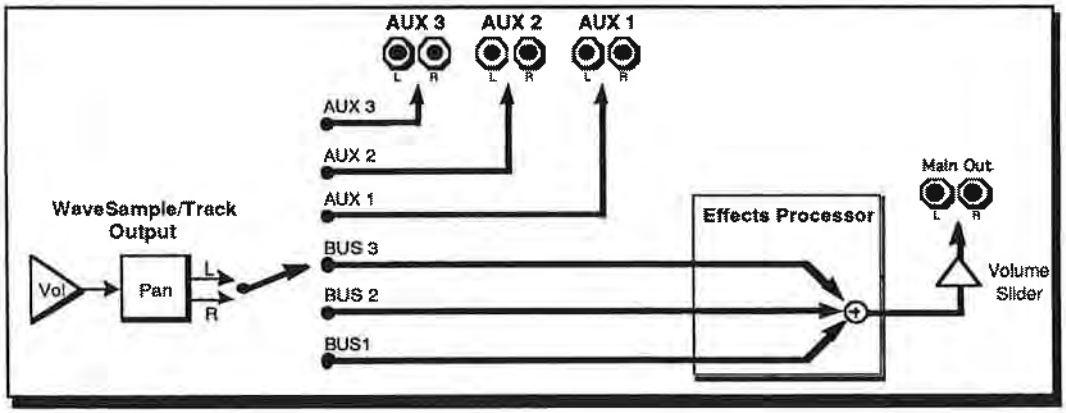
EDIT AMP	**BOOST**
	Press Edit / Amp / 6

Setting this parameter to ON gives the selected WaveSample a +12dB boost in volume. Be careful that adding boost to a sample doesn't cause it to become so loud that it starts to clip.

Range: OFF, ON

EDIT AMP	**OUT (Output Routing)**
	Press Edit / Amp / 9

This parameter enables you to assign instruments, layers, or WaveSamples to any of the three effects busses, or to any of the three auxiliary stereo outputs (AUX 1, 2, or 3). This diagram shows how voices are routed to the Effects Processor and to the Main and Auxiliary Output jacks.



Range: BUS 1, BUS 2, BUS 3, AUX 1, AUX 2, or AUX 3

Note: When Sample•Source Select/REC SRC Field 1 is set to INPUT+FX, all Voices go DRY, and any WaveSamples that were routed to BUS 1, 2, or 3 will all show OUT= DRY - - FX BUS IN USE on this screen.

ASR–10 Envelopes

Each ASR–10 WaveSample has three envelopes. These envelopes are hard-wired to the pitch, filter cutoff frequency, and amplitude of the WaveSample. Env 1 and Env 2 can also be assigned elsewhere, wherever a modulator is selectable. Each of the three envelope buttons is directly above the page button for the function it's associated with:

- Env 1 is routed to WaveSample pitch.
- Env 2 is routed to WaveSample filter cutoff frequency.
- Env 3 is routed to WaveSample amplitude (volume).

The indicator lights in the display tell you which one of the three Envelope pages you are on. For example when you are on the Edit/Env 2 page, the FILTER and ENV indicators are lit.

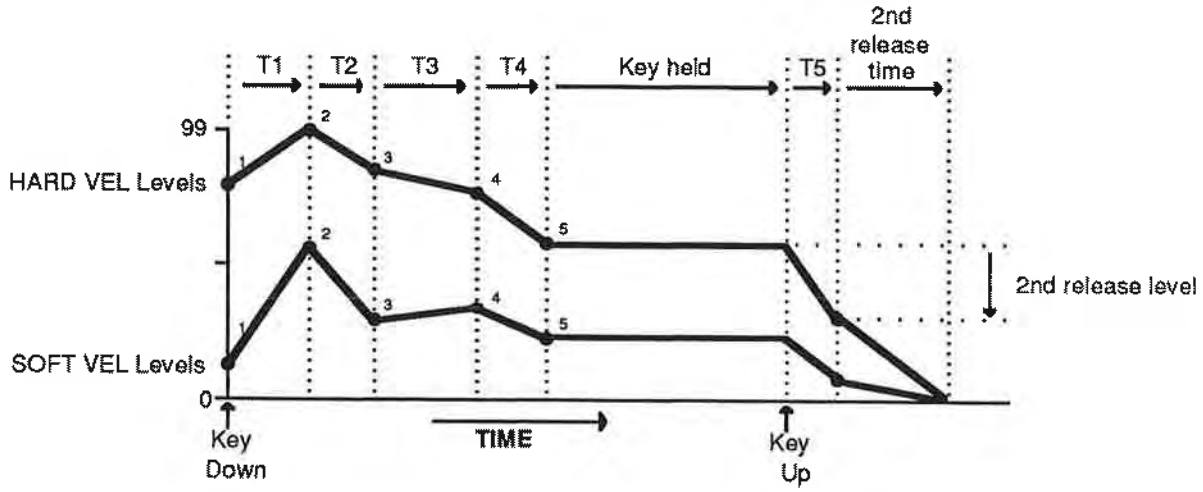
About ENV 3

In the ASR–10, the response of the WaveSample volume, ENV 3, and the Instrument mix level all use the same (non linear) response curve. As shown below, setting any of these volume controls to 50 will attenuate a sound by 16 dB.

HARD OR SOFT LEVEL RESPONSE (dB)	LEVEL AMPLITUDE
99	0 dB
75	-8 dB
50	-16 dB
25	-32 dB
6	-56 dB
0	-∞ (infinity)

Typical ASR–10 Envelope

The figure below shows a typical ASR–10 envelope shape:



The ASR–10 envelope is a descendant of the venerable ADSR (attack, decay, sustain, release) envelope, with a number of refinements. While the ADSR envelope gives you control over three times segments and one level, the ASR–10 envelopes have six times and six levels.

What's more, the envelope parameters allow you to describe two entirely different shapes; one for maximum velocity (HARD VEL) and one for minimum velocity (SOFT VEL). When a key is struck, the envelope generator actually *interpolates* between the two, creating a curve that is a combination of the two based on the incoming velocity.

Edit/Env 1, Env 2, Env 3 Pages

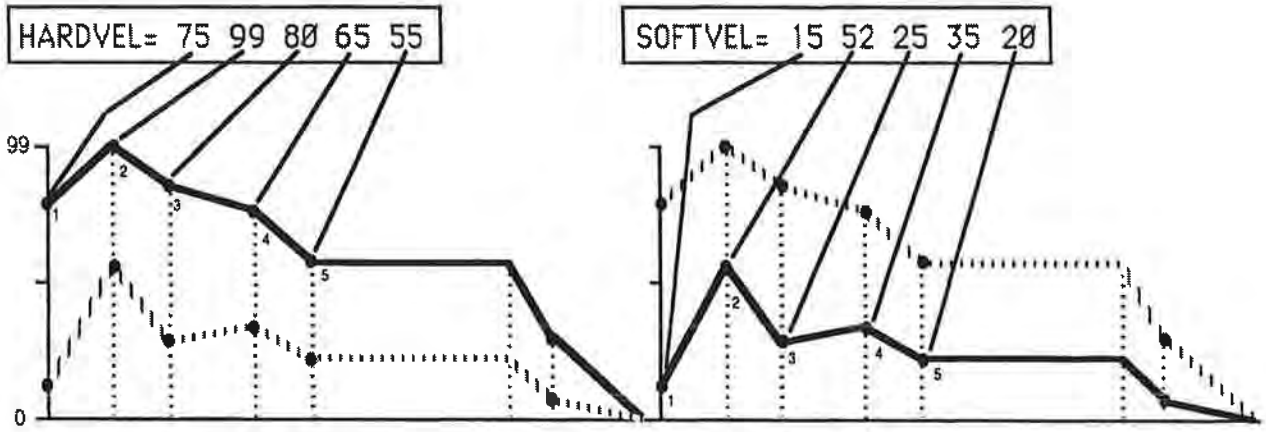
Each of the three Envelope pages (Edit/Env 1, Edit/Env 2, and Edit/Env 3) contain the following parameters. Use these parameters to adjust the envelopes.

EDIT ENV	**HARD VEL LEVELS**
	Press Edit / Env1, Env2, or Env3 / 1

HARDVEL shows the five levels for the envelope that will be used with maximum velocity. Level-1 is the initial level; in a traditional ADSR shape, Level-1 is always set to zero. Level-5 is the sustain level.

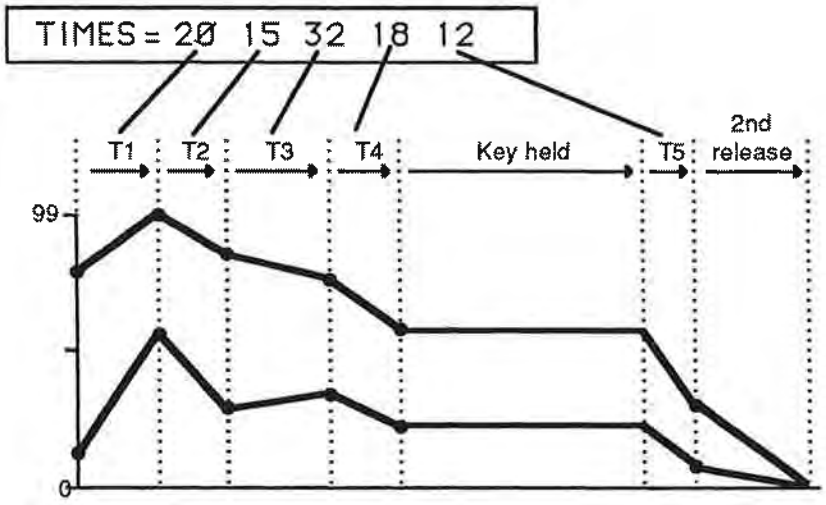
EDIT ENV	**SOFT VEL LEVELS**
	Press Edit / Env1, Env2, or Env3 / 2

SOFTVEL shows the five levels for the envelope that will be used with minimum velocity.



EDIT ENV	**TIMES**
	Press Edit / Env1, Env2, or Env3 / 3

TIMES shows five time parameters. The first four are the times between the five levels. The fifth time is the (first) release time.



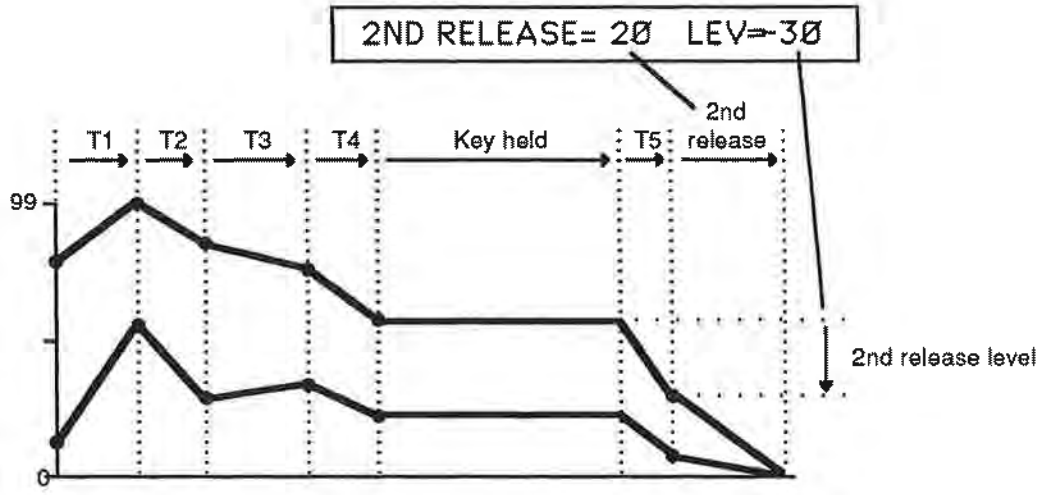
ENVELOPE TIMES

value	time (in sec)	value	time (in sec)	value	time (in sec)	value	time (in sec)	value	time (in sec)
0	.00	20	.20	40	.82	60	3.2	80	13
1	.01	21	.22	41	.88	61	3.5	81	14
2	.02	22	.23	42	.94	62	3.7	82	15
3	.03	23	.25	43	1.0	63	4.0	83	16
4	.04	24	.27	44	1.0	64	4.3	84	17
5	.06	25	.29	45	1.1	65	4.6	85	18
6	.07	26	.31	46	1.2	66	4.9	86	19
7	.08	27	.33	47	1.3	67	5.3	87	21
8	.08	28	.35	48	1.4	68	5.7	88	22
9	.09	29	.38	49	1.5	69	6.1	89	24
10	.10	30	.41	50	1.6	70	6.5	90	26
11	.11	31	.44	51	1.7	71	7.0	91	28
12	.11	32	.47	52	1.8	72	7.5	92	30
13	.12	33	.50	53	2.0	73	8.1	93	32
14	.13	34	.54	54	2.1	74	8.6	94	34
15	.14	35	.58	55	2.3	75	9.3	95	37
16	.15	36	.62	56	2.4	76	9.9	96	39
17	.16	37	.66	57	2.6	77	10	97	42
18	.17	38	.71	58	2.8	78	11	98	45
19	.19	39	.76	59	3.0	79	12	99	49

EDIT ENV	2ND RELEASE (TIME) - LEV
	Press Edit / Env1, Env2, or Env3 / 4

2ND RELEASE time and 2nd release LEVEL provide one more breakpoint in the envelope, which occurs *after* the release (T5). The LEVEL value controls where the breakpoint will be placed *relative to the current value of the envelope* at the time the key was released.

The following diagram illustrates how the 2nd RELEASE time and LEVEL affect both the HARDVEL and SOFTVEL levels of an envelope.



EDIT ENV	ATTACK TIME VEL AMT
	Press Edit / Env1, Env2, or Env3 / 5

ATTACK TIME VELOCITY AMOUNT makes TIME-1 shorter with increasing velocity. If TIME-1 is set to zero this parameter will have no effect.

Range: 0 to 99

EDIT ENV	KBD TIME SCALING
	Press Edit / Env1, Env2, or Env3 / 6

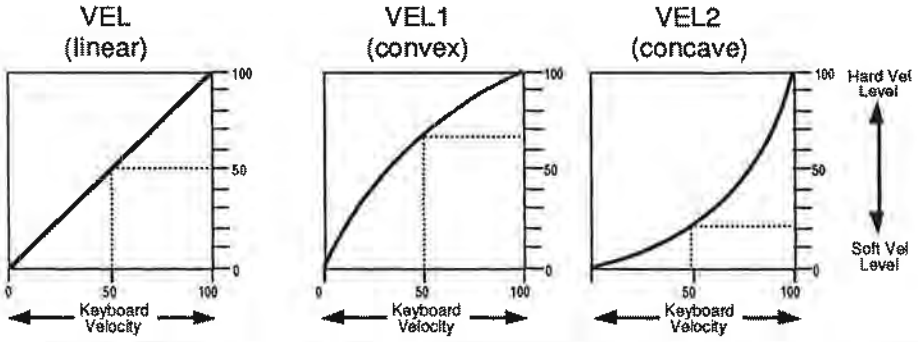
KBD TIME SCALING changes the envelope times above and below the root key. Positive values make the envelope shorter for keys above the root key, and longer for keys below the root key. Negative values have the opposite effect. The root key is not affected by this parameter. 2ND RELEASE time is also not affected by this parameter.

Range: -99 to +99

EDIT ENV	SOFT VEL CURVE
	Press Edit / Env1, Env2, or Env3 / 7

This parameter determines how incoming velocity will affect the envelope's interpolation between the SOFTVEL and HARDVEL level settings. There are four settings:

- OFF — The envelope uses only the HARDVEL levels, regardless of velocity.
- VEL, VEL1, or VEL2 — Incoming velocity will modulate the envelope levels between the SOFTVEL and HARDVEL level settings, according to one of the following velocity curves:



EDIT ENV	ENVELOPE MODE
	Press Edit / Env1, Env2, or Env3 / 8

ENVELOPE MODE has three states:

- NORMAL: as pictured in the previous envelope diagrams.
- FINISH: envelope runs its full cycle, ignoring the key-up event.
- REPEAT: In this mode, the envelope will run through levels 2 through 5, using times 1 through 4, repeatedly as long as the key is down.

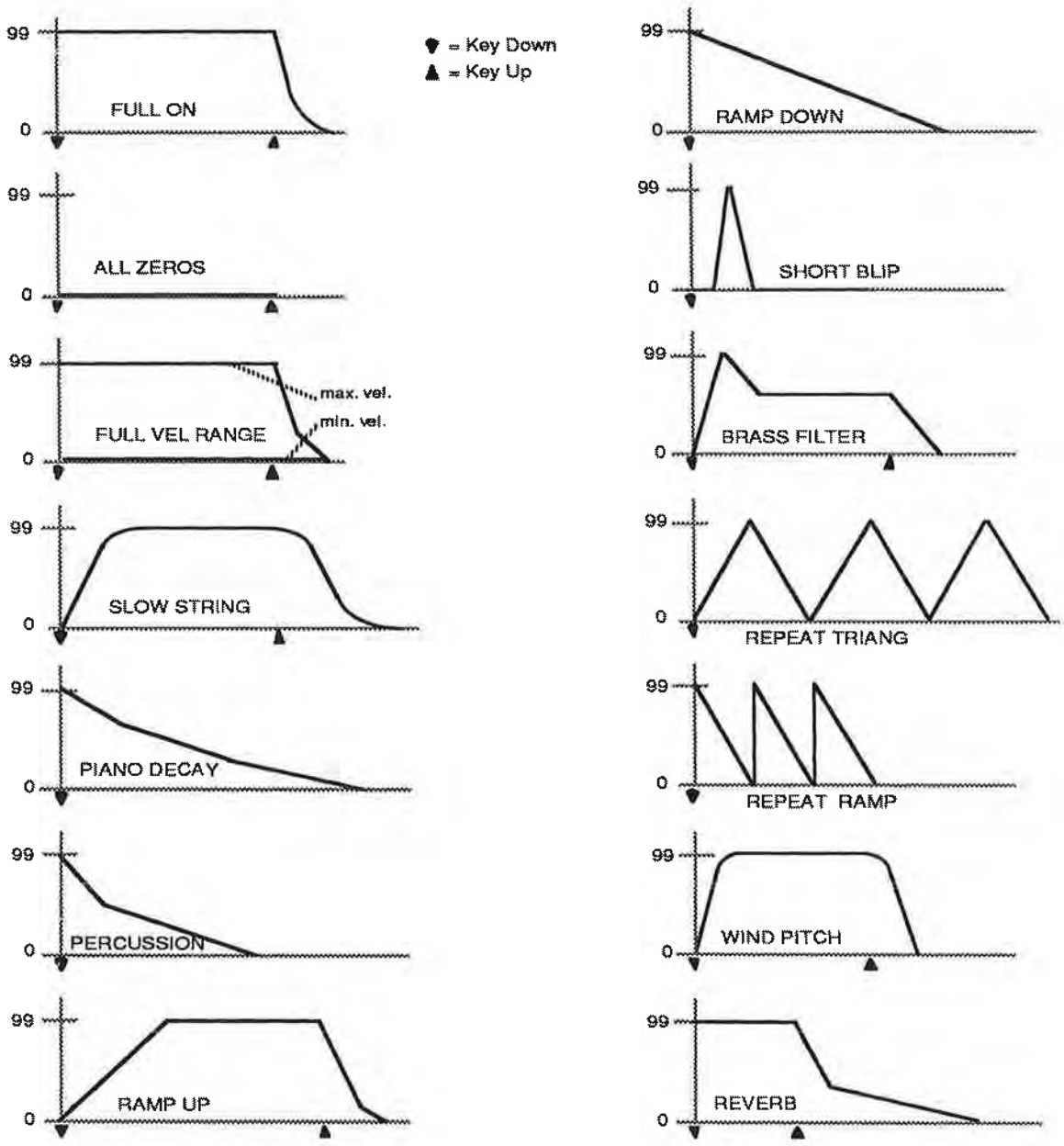
EDIT ENV	ENVELOPE (Envelope Template)
	Press Edit / Env1, Env2, or Env3 / 0

The last parameter on the Envelope page usually reads ENVELOPE=CURRENT VALUE. This is the envelope template parameter.

There are a number of "typical" envelope templates permanently stored in the ASR–10; they each have a name. When you select one of these envelopes, its twenty-one parameters are *copied* into the envelope of the WaveSample or WaveSamples you are editing. The old values are temporarily saved as the CURRENT VALUE. You can go back to CURRENT VALUE any time you wish, unless you change any of the values in one of the template envelopes. Editing individual parameters of *any* envelope causes it to become the CURRENT VALUE of that particular envelope.

The following diagram illustrates the 14 typical envelope templates.

Envelope Templates



Copying Envelopes

Pressing ***Enter•Yes*** while the envelope template parameter is displayed causes the current set of parameters to be saved in a special buffer. The message ENVELOPE PARAMS SAVED will appear briefly. There is only one SAVED envelope in the entire ASR–10 at any one time. This envelope becomes your own template, which can be selected on the envelope template page by selecting ENVELOPE=SAVED. You can use this function to copy an envelope from one WaveSample to another.

Edit/LFO Page

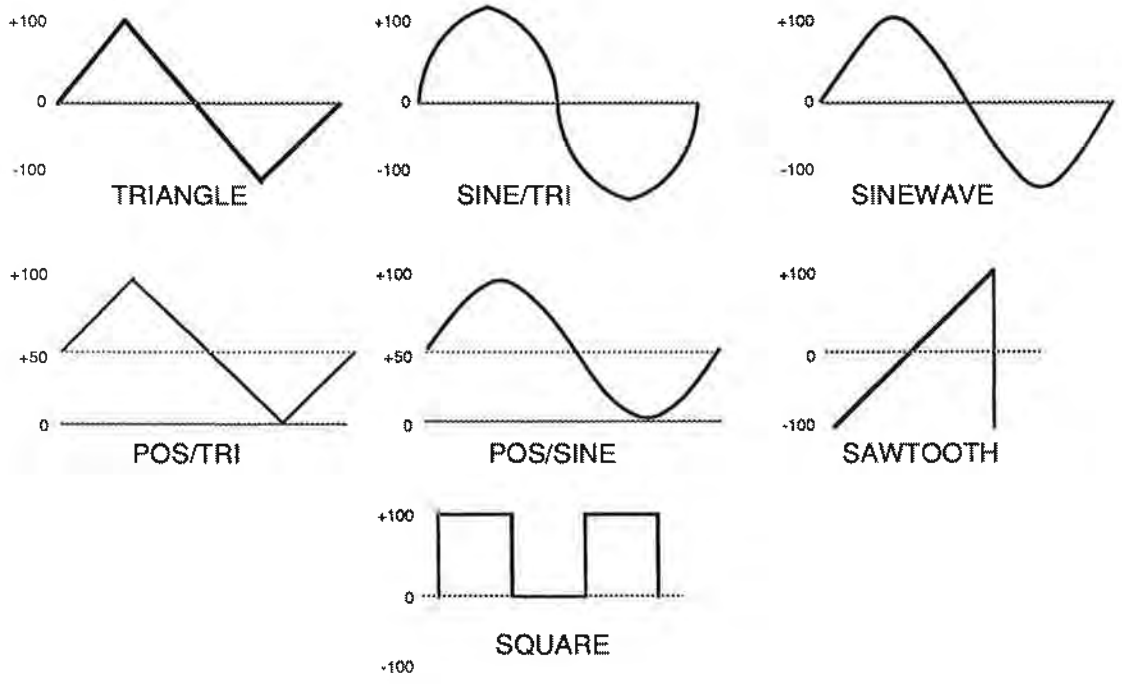
The parameters on this page control the Low Frequency Oscillator (LFO) that can be used to create vibrato, tremolo and other effects. To access these parameters, press ***Edit*** then ***LFO***.

EDIT LFO	**WAVE - RATE**
	Press Edit / LFO / 1 (for Wave) 2 (for Rate)

LFO Wave

Use this parameter to select one of the seven LFO waveforms:

- TRIANGLE — the triangle wave is commonly used to modulate pitch to produce vibrato.
- SINE/TRI — a mixture of a sine and triangle wave, a sort of pointy sine wave.
- SINEWAVE — the sine wave is a pure fundamental frequency, more rounded in its peaks and valleys than the triangle wave.
- POS/TRI — the POS/TRI is a positive-only triangle wave useful for simulating vibrato on instruments like the guitar, where vibrato techniques are limited to bending the note up.
- POS/SINE — the POS/SINE is a positive-only sine wave, similar in application to the POS/TRI.
- SAWTOOTH — the sawtooth wave is commonly used for special effects.
- SQUARE — this positive-only square wave is useful for producing in-tune trill effects.



LFO Rate

This parameter allows you to set the rate (or speed) of the LFO, making the LFO modulation faster or slower.

Range: 0 to 99

LFO FREQUENCIES

value	time (in Hz)	value	time (in Hz)	value	time (in Hz)	value	time (in Hz)	value	time (in Hz)
0	.05	20	1.25	40	6.25	60	11.25	80	16.25
1	.05	21	1.50	41	6.50	61	11.50	81	16.50
2	.06	22	1.75	42	6.75	62	11.75	82	16.75
3	.06	23	2.00	43	7.00	63	12.00	83	17.00
4	.06	24	2.25	44	7.25	64	12.25	84	17.25
5	.07	25	2.50	45	7.50	65	12.50	85	17.50
6	.07	26	2.75	46	7.75	66	12.75	86	17.75
7	.08	27	3.00	47	8.00	67	13.00	87	18.00
8	.08	28	3.25	48	8.25	68	13.25	88	18.25
9	.09	29	3.50	49	8.50	69	13.50	89	18.50
10	.10	30	3.75	50	8.75	70	13.75	90	18.75
11	.11	31	4.00	51	9.00	71	14.00	91	19.00
12	.13	32	4.25	52	9.25	72	14.25	92	19.25
13	.14	33	4.50	53	9.50	73	14.50	93	19.50
14	.17	34	4.75	54	9.75	74	14.75	94	19.75
15	.20	35	5.00	55	10.00	75	15.00	95	20.00
16	.25	36	5.25	56	10.25	76	15.25	96	20.25
17	.33	37	5.50	57	10.50	77	15.50	97	20.50
18	.50	38	5.75	58	10.75	78	15.75	98	20.75
19	1.0	39	6.00	59	11.00	79	16.00	99	21.00

EDIT LFO **LFO DEPTH - DELAY**

Press Edit / LFO / 3 (for Depth) 4 (for Delay)

LFO Depth

LFO DEPTH controls the initial amount of LFO. The effect of any LFO DEPTH modulator will be added to this amount.

Range: 0 to 99

LFO Delay

DELAY controls the time it takes for the LFO to go from zero to the amount set with the LFO DEPTH parameter. Values above 0 will cause the LFO to take longer to achieve its full depth. This is useful for creating delayed vibrato, tremolo, etc.

Range: 0 to 99

EDIT LFO	**LFO MODE**
	Press Edit / LFO / 5

Use this parameter to set the LFO in one of two modes:

- RESTART OFF — The LFO will cycle continuously without restarting.
- RESTART ON — The LFO waveform will return to the beginning of its cycle each time a new key is struck.

EDIT LFO	**DEPTH MOD - AMOUNT**
	Press Edit / LFO / 7

This parameter allows you to select a modulation source and the modulation amount for LFO DEPTH. The effect of LFO DEPTH MOD is *added* to the effect of the LFO DEPTH parameter. Either parameter can be used to generate LFO modulation. If you have the LFO DEPTH set to zero, you can still get LFO modulation by using any of the modulation sources on this page. Any of the fifteen available modulators can be applied to modulate LFO DEPTH.

EDIT LFO	**RATE MOD - AMOUNT**
	Press Edit / LFO / 8

This parameter allows you to select a modulation source and the modulation amount for the LFO RATE. The effect of RATE MOD is *added* to the effect of the LFO RATE parameter. Any of the fifteen available modulators can be applied to modulate the LFO RATE.

Edit/Layer Page

The parameters on this page control the layers within an instrument. To access them press ***Edit***, then ***Layer***.

EDIT LAYER	**LYR GLIDEMODE**
	Press Edit / Layer / 0

Use this parameter to create a portamento glide between notes. This is a monophonic effect that is triggered when you play legato notes. When set to PEDAL, you must play legato notes (or sustained notes) and depress the foot pedal to start the glide. The ASR–10 uses *constant-time* glide: it will take the same amount of time to glide between notes that are close together (i.e. C4 to D4) as it will to glide between notes that are far apart (i.e. C4 to C7). Glide time is determined by the LAYER GLIDETIME parameter.

For monophone voice assignment without glide, set LAYER GLIDEMODE to LEGATO, TRIGGER, or MINIMODE, and set the LAYER GLIDETIME to zero.

- OFF — Glide is off. This is normal polyphonic operation with no portamento.
- LEGATO — Enables monophonic voice assignment, allowing only one voice to be active in the layer. If a key is depressed before another is released, the voice will glide from the old note to the new one. The envelopes and WaveSamples are retriggered or restarted each time there is a new key down.
- PEDAL — Enables the sustain pedal to start the polyphonic glide. In this mode, the glide and polyphony are controlled by the sustain pedal. When the pedal is up, there is no glide and the layer is fully polyphonic. When the sustain pedal is depressed, the glide is enabled, and the polyphony is limited to the number of notes being sounded at the moment. For example, with the pedal up play a three-note chord. Press the sustain pedal and lift your fingers off the keys (the three notes will sustain). Now press another key: one of these sustaining notes will glide to the new note. The envelopes and WaveSamples are retriggered, or restarted, each time there is a new key down. Keeping a key down will prevent the note from being stolen. This allows controlled voice leading.
- TRIGGER — This mode plays monophonically, with the envelopes and WaveSamples retriggering, or restarting, each time there is a new key down. This mode works just like LEGATO mode, except that it has no "note memory" (see below). It is ideal for adding monophonic piano "thuds" to piano instruments.
- MINIMODE — Similar to mono mode on old analog synths. The voice will play with one-voice polyphony, and new notes will glide from the previous note if the previous note is still held. Unlike the LEGATO, PEDAL, or TRIGGER modes, the envelopes and WaveSamples are only restricted from the beginning when you play staccato—play a new note with no other keys down.

Note: LEGATO and MINIMODE feature "note memory." If you release a key while still holding down another key, the pitch will glide back to the note you are holding. Holding down the sustain pedal will dynamically disable note memory.

GLIDE MODES

Mode	Staccato Glide	Legato Glide	Legato Retrigger	Note Memory
LEGATO	NO	YES	YES	YES
PEDAL (with pedal down)	YES	YES	YES	NO
TRIGGER	NO	YES	YES	NO
MINIMODE	NO	YES	NO	YES

EDIT LAYER

LAYER GLIDETIME

Press Edit / Layer / 1

Use this parameter to set the speed of the glide between notes. Lower values yield a faster glide; higher numbers, a slower glide.

Range: 0 to 99

EDIT LAYER

LEGATO LAYER

Press Edit / Layer / 2

The legato layer is an ENSONIQ innovation that enables the ASR–10 to capture the elusive nuances of acoustic instruments. The sound of a wind instrument, the flute for example, is different when it is "tongued" than when notes are played in a connected manner (called legato). When two notes are "slurred" or "trilled" the characteristic attack of the second note is not present. The ASR–10 allows a simulation of this effect when used with one of the glide modes.

The rules for glide, described above, determine when two notes will be connected. The glide time determines how fast it will get to the new note. The legato layer parameter selects which layer, and therefore what sound, will play on the new note.

For example, suppose that layer one is a flute sample with a tongued attack, and layer two is a flute sustain without any attack. To use this function:

On the Edit/Instrument page:
- Enable both layer 1 and layer 2 in the patch. The display reads 00 PATCH=1 2 - - - - - -.
- Scroll right, enable layer 1, the attack, as a keydown layer. Disable layer two as a keydown layer. The display reads KEYDWN LAYERS=1 - - - - - - -.

On the Edit/Layer page for layer 1:
- Set GLIDEMODE=LEGATO.
- Set LEGATO layer=2.
- Adjust GLIDETIME as you see fit.

If a key is depressed before another is released, the new note will play the WaveSample in the Legato layer (layer 2).

EDIT LAYER	**LYR VEL LO - HI**
	Press Edit / Layer / 3

This parameter allows you to set the velocity range in which the layer will respond. Outside the range selected, the layer will not play.

Range: 0 to 127

EDIT LAYER	**PITCH TBL**
	Press Edit / Layer / 4

This parameter determines whether a layer will use the STANDARD (equal tempered) pitch table, or play each WaveSample in the layer at the pitch it would play on C4 (NO PITCH), or use any pitch table you create. See the discussion of the Command/Pitch EDIT PITCH TABLE command earlier in this section for more on creating and editing pitch tables.

Range: STANDARD, NO PITCH, USER PITCH TABLES 1-8.

EDIT LAYER	**LYR NAME**
	Press Edit / Layer / 5

Use the data entry controls to change the name of an existing layer, or name a new one that you've just created (the ***Left/Right Arrow*** buttons move the cursor, and the ***Data Entry Slider*** and the ***Up/Down Arrow*** buttons change the characters).

EDIT LAYER	**DELAY - VEL AMOUNT**
	Press Edit / Layer / 6 (for Delay) 7 (for Velocity Amount)

The DELAY parameter determines how long voices within the selected layer will wait after a key is struck before playing. This parameter is adjustable from 0 to 5000, with each value increasing the delay by 1 millisecond.

The VELOCITY AMOUNT is a modulation amount, with velocity serving as a dedicated modulation source. Positive values will cause harder velocities to increase the DELAY time. Negative values will cause harder velocities to decrease the DELAY time. Setting the DELAY to zero will disable velocity modulation of DELAY time.

EDIT LAYER	**LAYER RESTRIKE TIME**
	Press Edit / Layer / 8

This parameter sets the amount of decay voices within the selected layer will have after they have been restruck. The higher the value, the longer the decay time.

Range: 0 to 99

EDIT LAYER	**STEREO LAYER LINK**
	Press Edit / Layer / 9

Stereo samples are stored in pairs of adjacent layers (1&2, 3&4, 5&6, or 7&8), with the LEFT channel always stored in an odd numbered layer (1, 3, 5 or 7), and the RIGHT channel stored across the same key range in the next higher even numbered layer. These paired adjacent layers are called *companion* layers.

The STEREO LAYER LINK parameter is automatically turned ON after a stereo sample is recorded. When this parameter is turned ON in the selected LAYER, it is simultaneously turned ON in the companion layer. When STEREO LAYER LINK= ON, simultaneous editing of both channels of a Stereo Sample is enabled (see below).

When STEREO LAYER LINK= ON, editing layer or WaveSample parameters in one layer will edit the same parameter in the companion layer. Editing with the ***Data Entry Slider*** will set the current parameter in both layers to the same absolute value. Editing with the ***Up/Down Arrow*** buttons will edit parameter values in both layers *relative* to one another.

When STEREO LAYER LINK= ON, all pitch modulation in the right layer will be slaved to the settings in the WaveSamples assigned to the same or overlapping key zones in the left layer. This includes BEND RANGE, NOISE RATE, and all pitch modulation. Pitch modulation settings for WaveSamples in the right layer will be ignored.

STEREO LAYER LINK can be turned OFF to allow independent editing of layers that contain one channel of a stereo sample.

If the companion layer does not exist, STEREO LAYER LINK cannot be turned ON in the selected layer. The screen will display NO COMPANION LAYER.

When performing wave data commands with STEREO LAYER LINK= ON, both the selected WaveSample and the WaveSample in the companion layer must be the same size. If the two WaveSamples are not the same size, the display will read SAMPLES NOT SAME SIZE, and the command will not be performed.

Hint: You can quickly access this parameter by double-clicking the ***Layer*** button when in Edit mode.

Range: OFF, ON

Command/Layer Page

These commands allow you to create, copy, and delete whole layers.

CMD LAYER	CREATE NEW LAYER
	Press Command / Layer / 0

Use this command to create a new layer in the currently selected instrument.

- Select CREATE NEW LAYER.
- Press ***Enter•Yes***. The display reads LAYER ## CREATED.

CMD LAYER	COPY LAYER
	Press Command / Layer / 1

Use this command to copy an existing layer and all the WaveSamples in it, either within an instrument or from one instrument to another.

- Select COPY LAYER. The display reads TO INST=UNNAMED INST.
- Select the destination instrument. Press ***Enter•Yes***.
- Select COPY=PARAMS ONLY/PARAMS+DATA. It is a good idea to choose PARAMS ONLY when copying a layer within the same instrument, allowing the wave data to be shared. This will conserve memory. However, *always* choose PARAMS+DATA when copying a layer from one instrument to another, or else the destination instrument will not contain the wave data when it is saved to disk, and subsequently reloaded.
- Press ***Enter•Yes***. The display reads LAYER ## CREATED.

Note: When STEREO LAYER LINK= ON, this command will also copy the companion layer. However, in order for this to work properly, there *must* be two undefined adjacent layers available.

CMD LAYER	DELETE LAYER
	Press Command / Layer / 2

Use this command to delete a layer in the currently selected instrument. This will delete all WaveSamples in the selected layer as well.

- Choose the layer you want to delete.
- Select DELETE LAYER.
- Press ***Enter•Yes***. The display reads DELETE LAYER ##.
- Press ***Enter•Yes*** or ***Cancel•No***.

Note: If you get the message NO EDIT LAYER SELECTED, simply press ***Edit*** and select the layer you want to copy or delete, and then proceed with the command.

Sequencer & Audio Track Concepts

Section 12 — Sequencer and Audio Track Concepts

This section contains an introduction to the ASR–10 sequencer and all the information you'll need to get started sequencing. For more detailed descriptions of the actual parameters, refer to the following section.

Sequencing on the ASR–10

The ASR–10 sequencer incorporates a range of features and capabilities you would expect to find in stand-alone or computer sequencers, yet with the advantage of being part of an integrated system.

The ASR–10 is both powerful and easy to use — having your sampler, and sequencer right at your fingertips in one unit is what makes the ENSONIQ approach to digital sequencing so intuitive and efficient.

If you can't wait to start sequencing, you can turn right to "Recording a Sequence" later in this section. We recommend, however, that you come back and familiarize yourself with the many other sequencer controls and functions described in this section. This is the only way to truly take advantage of the power of the ASR–10 sequencer.

Digital Sequencing

Imagine a recorder which, instead of recording the sounds of an instrument, records the same kind of digital information that is sent and received via MIDI — key down, key up, key number and velocity, pitch bend, mod wheel, program changes and so on — and you have imagined a *digital sequencer*.

A sequencer records and plays back the "control information" rather than the actual notes. This means that there is no degradation of the sound in the recording process no matter how many times you overdub or rerecord a part. A sequencer is sort of like an electronic player piano.

It is important to bear in mind that a sequencer only records what you play. Sequencer memory is used up on the basis of *Events* (keys struck, controllers, etc.), while a tape recorder's memory (the tape) is always used up by the same amount over a fixed period of time.

This means that a sequencer will use virtually the same amount of memory to record 100 notes, whether you play those notes over ten seconds or ten minutes. When you strike a key, the sequencer records a *Key Down* event. It then counts the *clock* pulses until you release the key, when it records a *Key Up* event. The amount of time between the key down and the key up doesn't really affect the amount of memory required to record the note. Compare this to an audio tape recorder. With tape, *time* is the main factor. A tape recorder will use the same amount of tape to record a minute of music, whether the signal contains one note or one hundred.

You might say that tape is *linear* — it is used at a fixed rate, while digital sequencer memory is *dynamic* — it is used only as needed. Understanding the difference will help you to manage the ASR–10 sequencer memory. For example, while key events (the notes you play) use up relatively little memory each, controllers such as pressure, mod wheel, pitch bend, etc., are recorded as a flood of numbers that can fill up the memory in a hurry. Thus, if you're trying to squeeze one more track into a sequence when there is not much memory left, you should go easy on the controllers.

What is a Sequence?

A sequence on the ASR–10 is a collection of eight independent tracks and their associated notes, controllers, etc.

The ASR–10 can have up to 80 sequences in its internal memory at once. Each sequence has its own 12 character name and time signature. A sequence can be up to 999 bars long.

You select the current sequence (the sequence you want to play or record) on the Edit/Seq•Song page. Press ***Edit***, then double-click on ***Seq•Song*** to go directly to the Select Seq/Song parameter.

You can save an individual sequence to disk, though in most cases it is more useful to save the entire song with all its related sequences.

What is a Song?

The sequences in memory can be chained together into a song. There can only be one song in memory at a time. The song also has its own 12 character name.

The song is selected on the Edit/Seq•Song page just as you would select a sequence. When the song is selected, the SONG indicator lights on the display.

In Song mode, sequences are assigned to play consecutively in any order, with up to 99 *Steps*, and up to 63 *Repetitions* of each step. Within each song step, individual tracks within the sequence can be muted or transposed.

After chaining together sequences into a song, you can record another complete set of song-length tracks that are separate from the individual sequence tracks, but share the same instruments. These "Song Tracks" are stored with the song. This makes the ASR–10 sequencer a sixteen track recorder (see Song Mode later in this section).

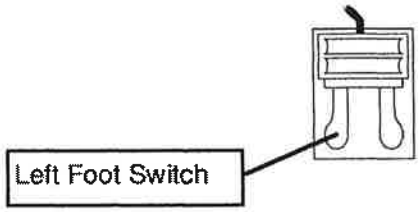
When you save a song to disk (using the SAVE SONG + ALL SEQS command on the Command/Seq•Song page), the ASR–10 automatically saves all the sequences currently in memory along with the song (it doesn't save the instruments; they must be saved separately). When you later load the song into memory (by itself or as part of a Bank), all the sequences will be loaded in along with it.

Banks

The way to make sure that all the instruments are loaded into the proper locations, all the mix and pan levels for each track are properly preserved, and that the proper effect and related settings are in place is to save the whole contents of the ASR–10 memory as a Bank. Loading a bank file (instead of loading the song and the various instruments separately) ensures that every instrument will be loaded into the proper ***Instrument•Sequence Track*** location, so that the tracks will all play the right sounds.

Sequencer "Transport Controls"

The three buttons to the right of the display serve to start and stop the sequencer, and to put it into Record and Overdub modes. In addition to these three buttons, the Left Foot Switch (available only when an optional SW-10 Dual Pedal Foot Switch is connected to the ASR–10 Foot Switch jack and the Edit/System•MIDI LEFT FOOT SW= STOP/CONT) can be used to start and stop the sequencer when both hands are busy.



- Pressing ***Play*** will start the current sequence or the song (whichever is selected) playing from the beginning.
- Pressing ***Stop•Continue*** will stop the sequencer (if pressed while it is running); or will play the current sequence or the song from wherever it was last stopped (if pressed while the sequencer is stopped).
- The Left Foot Switch will duplicate the behavior of the ***Stop•Continue*** button.

- Pressing ***Play*** while holding down ***Record*** will start the sequencer recording on the current track from the beginning of the sequence or song.
- Pressing ***Stop•Continue*** while holding down ***Record*** will start the sequencer recording on the current track from wherever it was last stopped.

- Pressing ***Record*** while the sequencer is playing will put the sequencer into "Punch In" mode. It will wait for you to either start playing, or press the Left Foot Switch before going into Record on the current track.

Sequencer Status

The sequencer is always in one of the following states; the sequencer Status Indicators in the upper right part of the display always show you the current status:

• STOP — Sequencer at rest

• PLAY — Playing current Sequence or the Song

• REC — Recording on current Track, first pass

• ODUB — Overdub: Re-recording current Track

• PLAY (flashing) — Countoff playing prior to going into Play, Record or Overdub

• REC (flashing) — Record Standby: waiting for you to play before going into Record (first Track only)

• ODUB (flashing) — "Punch-in" Standby: waiting for you to either play or press the Left Foot Switch before going into Overdub

Loading and Saving Sequencer Data

There are two ways that ASR–10 Sequencer data can be stored on a disk:

- **SONG File.** A song file contains a song *and* all its related sequences. Loading a song file from disk will completely erase the current contents of the ASR–10 sequencer memory, replacing whatever is there with the song and sequences from the disk file.
- **SINGLE SEQUENCE File.** This type of file contains just one sequence. Loading a single sequence file will *not* erase the sequencer data already in memory — the new sequence simply becomes one more sequence in the internal memory which can be selected or used as a step in a song. There can be up to 80 sequences in memory at once.

Loading a Song or a Single Sequence from Disk

Both types of sequencer files are accessed from the Load/Seq•Song page.

- Press ***Load***, then press ***Seq•Song.***
  The LOAD indicator flashes, meaning that the ASR–10 is showing you disk files.
- Press the ***Up*** or ***Down Arrow*** button until the file you want to Load is showing on the display:

When a SONG file is showing, the SONG indicator lights next to the SEQ indicator.

When a single Sequence file is showing, only the SEQ indicator lights.

- Press ***Enter•Yes*** to load the sequencer file showing on the display. After you have loaded a song or sequence, it is automatically selected.

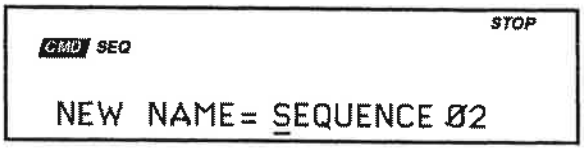
Important:

Remember that loading a song or sequence does not ensure that the right instruments are loaded into the proper ***Instrument•Sequence Track*** locations. Each track of each sequence will play whatever instrument is in that location at the time. The way to make sure everything is in the right place is to save the contents of memory (including the song) as a bank.

Saving a Single Sequence to Disk

Use the SAVE CURRENT SEQUENCE command to save a single sequence.

- On the Edit/Seq•Song page, select the sequence you want to save.
- Insert a formatted disk into the drive.
- Press ***Command***, then press ***Seq•Song.***
- Press the ***Left*** or ***Right Arrow*** button until the display reads SAVE CURRENT SEQUENCE.
- Press ***Enter•Yes.***
- Edit the Sequence Name (if needed):



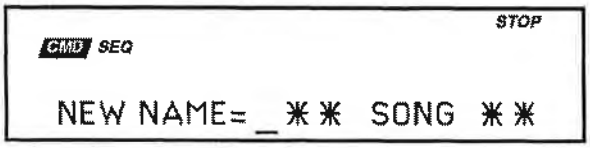
The display shows the current name of the sequence, with a cursor (underline) beneath the first character. If you want to give the sequence a new name, do so at this time. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the underlined character, then press the ***Left/Right Arrow*** buttons to move the underline. Repeat until the display shows the name you want (if you don't need to rename the sequence, just skip this step).

- Press ***Enter•Yes***. The display will read SAVING <FILE NAME> while the sequence is being saved.
- If there is already a sequence file with the same name on the disk, the display will ask DELETE OLD VERSION? Press ***Enter•Yes*** to save the sequence, replacing the one on the disk. Or press ***Cancel•No*** to abort the procedure.
- If there is not enough free space on the disk, the display will say NOT ENOUGH DISK SPACE. Save the sequence to another disk (or delete some files from the disk).
- If you try to SAVE CURRENT SEQUENCE while the song is selected, the ASR–10 will not execute the command, responding USE SAVE SONG + ALL. Again, make sure the sequence is selected before trying to save it.

Saving a Song (along with all Sequences) to Disk

Once you have created a song or made changes to an existing one, you can save the song to a formatted ASR–10 disk. In addition to saving the song itself, the SAVE SONG + ALL SEQS command saves all the individual sequences currently in memory (whether they are part of the song or not). To save a song:

- Insert a formatted disk into the drive.
- Press ***Command***, then press ***Seq•Song.***
- Press the ***Left*** or ***Right Arrow*** button until the display reads SAVE SONG + ALL SEQS.
- Press ***Enter•Yes***.
- Edit the song name (if needed):

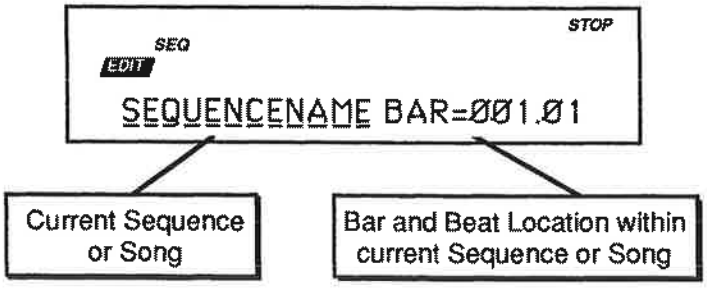


The display shows the current name of the song, with a cursor (underline) beneath the first character. If you want to give the song a new name, do so at this time. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the underlined character, then press the ***Left/Right Arrow*** buttons to move the underline. Repeat until the display shows the name you want (if you don't need to rename the song, just skip this step).

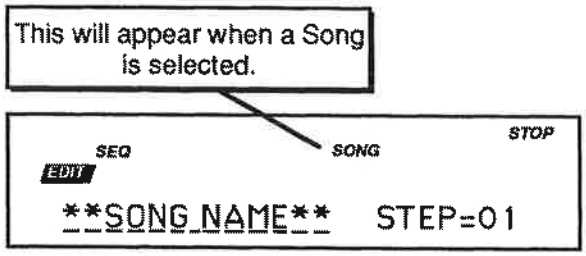
- Press ***Enter•Yes***. The display will read SAVING <SONG NAME> while the song is being saved.
- If there is already a song file with the same name on the disk, the display will ask DELETE OLD VERSION? Press ***Enter•Yes*** to save the song, replacing the one on the disk. This is for updating songs to which you have made changes. Or press ***Cancel•No*** to abort the procedure.
- If there is not enough free space on the disk, the display will say NOT ENOUGH DISK SPACE. Save the song to another disk (or delete some files from the disk).
- Note that the song and all sequences in memory are saved as one file. You cannot extract a single sequence from a song file and load it separately. If you want access to a sequence individually, you must first load the song into the internal memory, and then save the individual sequence using the SAVE CURRENT SEQUENCE command.

Selecting a Sequence/Song

Press the ***Edit*** button, then the ***Seq•Song*** button. The following screen will appear:



This screen is where you select a sequence or the song. With the selected sequence or song name underlined (as shown above), use the ***Up/Down Arrow*** buttons or the ***Data Entry Slider*** to select a different sequence or song. The display will indicates whether a song is selected:



Hint: You can always get back to the Current Sequence/Song screen by double-clicking the ***Seq•Song*** button (when in Edit mode).

Playing a Sequence/Song

Try selecting a sequence, and pressing the ***Play*** button in the sequencer section, to the right of the display. The selected sequence will begin to play.

While one sequence is playing you can select another one. You will see the name of the new sequence in the display, but the original one will continue to play. When the first sequence is finished, the sequencer will switch to the new sequence, and it will play. In this fashion you can string sequences together in real time, as they play.

Press the ***Stop•Continue*** button or the Left Foot Switch to stop the current sequence.

Sequence Tracks

Each sequence is composed of up to eight tracks, which correspond to the eight ***Instrument•Sequence Track*** locations. That is, each instrument location also doubles as a track. Whatever is recorded on a given track will play on the instrument that is loaded into that ***Instrument•Sequence Track*** location.

You can only record on one track at a time. All sequence recording is done in Edit mode, and in Edit mode you can only select one ***Instrument•Sequence Track*** at a time.

You "mix down" sequences on the Edit/Track page. From the screens on this page you can play, mute, or solo each track; you can also adjust the MIX level, PAN setting, and Output Bus for each track (MIX, PAN, and OUT for each track are remembered when you save a bank).

The Audio Tracks also allow you to adjust the MIX level, PAN setting, and Output Bus for each track.

Note: When Editing Tracks, remember that the selected Track's (***Instrument•Sequence Track*** or ***Audio Track*** with its LED solidly lit yellow) parameters are the ones that are being viewed.

Creating a New Sequence

- Press ***Command***, then double-click the ***Seq•Song*** button. The display shows CREATE NEW SEQUENCE.
- Press ***Enter•Yes***. The display will show NEW NAME= SEQUENCE 02 (or 03, 04, etc.). There is a cursor beneath the first character. You can now give the sequence a name of your choosing, or use the default name by skipping the next step.
- Name the new sequence, using the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change each character, and then pressing the ***Left/Right Arrow*** buttons to move the cursor to the next character. Continue this until the display shows the name you want.
- Press ***Enter•Yes***.
- Select the time signature for the new sequence (or, if the time signature that's showing is the one you want, just press ***Enter•Yes*** twice). The time signature for the sequence is set at this time and *cannot be changed later*. Adjust the top half of the fraction (numerator), then scroll right to select the bottom half of the fraction (denominator) and adjust that.
- Press ***Enter•Yes***. The new sequence is created and becomes the currently selected sequence.

Recording a Sequence

Here we will describe recording a new sequence from scratch. First we will concentrate on sequencing with the ASR–10 alone, and then cover recording MIDI instruments.

1) Create a New Sequence:

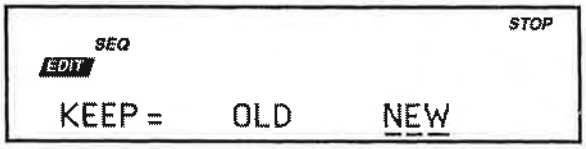
- Following the steps outlined earlier in this section, and create a new sequence.

2) Record the First Track:

- Connect a MIDI keyboard or other MIDI Controller to the ASR–10's MIDI In jack.
- Select any loaded instrument by pressing its ***Instrument•Sequence Track*** button. This makes it the current track.
- While holding down ***Record***, press ***Play***. The ASR–10 immediately puts you on the Edit/Seq•Song page and the click track starts playing (assuming CLICK=ON or REC on the Edit/Seq•Song page), at the current tempo. The first beat of each measure is emphasized,
- Adjust the tempo. Press the ***Right Arrow*** button three times to scroll to the TEMPO parameter. Adjust it to the tempo you want with the ***Data Entry Slider***, or press the ***Enter•Yes*** button twice in a row to use the tap tempo function.
- Play the keyboard to start recording. The bar in which you start playing becomes bar 1 of the sequence.
- Press ***Stop•Continue*** to end recording. The display will say XXX BARS - KEEP TRACK? The length (in bars) of the first track determines the length of the sequence.
- Press ***Enter•Yes*** to keep the track, defining the length of the sequence, or
- Press ***Cancel•No*** to erase the first track and start over again.

3) Record Additional Tracks:

- Select a different track. Again, selecting any loaded instrument by pressing its ***Instrument•Sequence Track*** button makes it the current track. Or you can leave the original ***Instrument•Sequence Track*** selected and overdub (re-record) that track.
- While holding down ***Record***, press ***Play***. After a one bar count off (if COUNTOFF=CLICK on the Edit/Seq•Song page) the ASR–10 goes into record and will record what you play on the selected track.
- Press ***Stop•Continue***. The display shows:



This is called the AUDITION page. While playing the sequence, press the ***Left/Right Arrow*** buttons to move the cursor between OLD and NEW to hear the original track or audition the new track.

- With **NEW** underlined, pressing ***Play*** will play the new track, and pressing ***Enter•Yes*** will KEEP the new track, erasing the original. When **OLD** is underlined, pressing ***Play*** will play the ORIGINAL track, and pressing ***Enter•Yes*** will KEEP the original track. Note that on this page, pressing ***Cancel•No*** will *always* keep the original track.

"Punching In" on a Track

Suppose you have a track that is fine for the first four bars but the next four bars need to be redone. You can "Punch In" at any point in the track by doing the following:

- Select the track you want to record on.
- Press ***Play*** to start the sequence or song playing.
- Press ***Record***. This puts the ASR–10 in Overdub Standby — the ODUB indicator flashes on the display and the sequencer is waiting for you to play keys before going into record.
- Start playing (or press the Left Foot Switch) at the point where you want to punch in. As soon as you play anything, the ASR–10 goes into Overdub (or Record for a new track) and records what you play, leaving intact the part of the track before the punch in.
- Press ***Stop•Continue***. You will see the Audition page as shown above, letting you play the new or the old track before deciding which to keep.

Recording the Default Track MIX Setting

Each ***Instrument•Sequence Track*** has its own default MIX setting that is recorded each time that a sequence track is recorded/overdubbed. This default MIX setting is fixed, and applies to the entire duration of the track (unlike *continuous* volume changes recorded with the Foot Pedal, or MIXDOWN VOLUME events recorded on the songs tracks — see later in this Section). If the Edit/Track MIX value is edited during playback, the default track MIX setting will be restored each time the sequence loops around. To permanently change the default MIX setting for a track:

- Set SEQ REC MODE= ADD on the Edit/Seq•Song page. This will prevent any unwanted changes to existing track events.
- Select the ***Instrument•Sequence Track*** for which you want to adjust the MIX.
- Press ***Play***, and then press ***Edit***, followed by ***Track***, and adjust the MIX value while listening to the track until you find the proper level.
- Once you've decided on the MIX value, press ***Stop•Continue***.
- While holding down ***Record***, press ***Play***. After the first bar of the sequence has been recorded, press ***Stop•Continue***.
- The KEEP= OLD/NEW Audition page will be displayed. You can't audition the volume change on the KEEP=OLD/NEW page. Select KEEP=NEW and press ***Enter•Yes*** to keep the results. Press ***Play***. You will now hear the new default MIX setting, and it will now be permanently recorded on the track. If you don't like the new default MIX setting, simply repeat the above procedure and set a new MIX level.
- It's a good idea to return to the Edit/Seq•Song page to set the SEQ REC MODE back to REPLACE for later sequencing.

Hint: You can use the optional CVP-1 Foot Pedal to record *continuous* volume changes onto a sequence track. Set PEDAL=VOLUME on the Edit/System•MIDI page, and then record the Pedal•CV controlled volume changes with SEQ REC MODE= ADD. This will record volume (MIDI Controller #7) events onto the track, which can then be edited like any other sequencer event.

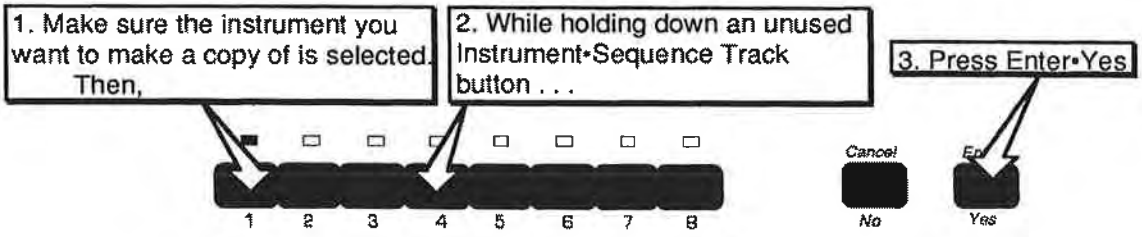
For information about mixing down song tracks, see **Recording MIXDOWN VOLUME and MIXDOWN PAN**, later in this Section.

Recording Another Track with a Copy of the Same Instrument

You will sometimes find that you want to record a second track using the same instrument. There is a quick way to make a "working copy" of any ASR–10 instrument in an unused location, without copying the sample data (thus conserving memory).

Suppose, for example, that you have recorded a track with a piano sound on Track 1. Now you want to record another track with the same sound, but with independent controllers, MIX level, PAN, OUT bus, etc. You can copy the piano into any empty ***Instrument•Sequence Track*** location (let's say Track 4) by following these three steps:

To Copy an Instrument to Another Location:



The display will read PIANO COPIED. ***Instrument•Sequence Track 4*** now contains a copy of the piano in ***Instrument•Sequence Track 1***. You can now record on this track just as you would any other. Keys, controllers, etc. recorded on the new track will be independent from any recorded on ***Instrument•Sequence Track 1***. You can Solo, Mute, MIX and PAN the new track independently from the original. The new track can be assigned to its own MULTI-IN MIDI CHANNEL and OUT bus. The two tracks simply "point to" the same sound. You can make as many copies of an instrument as you want. Instrument copies will not take up much additional memory, because they contain only "pointers" to the original sound.

Additional Notes

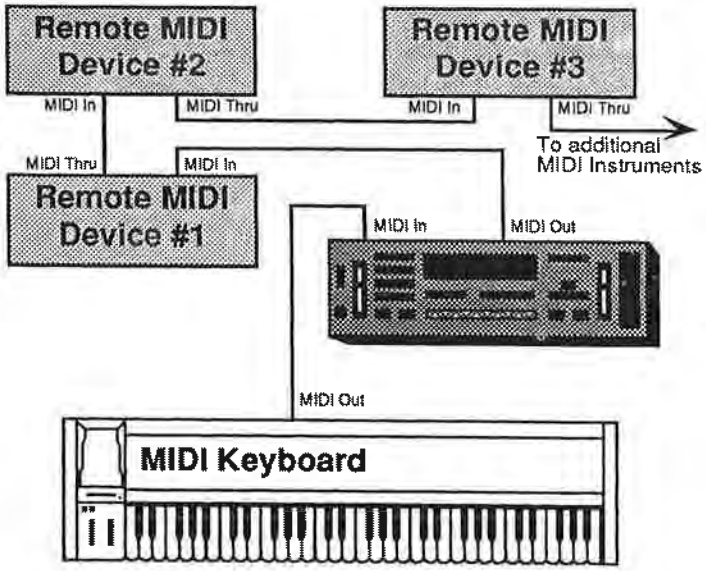
- When you select an ***Instrument•Sequence Track*** location that contains a copy, it looks just like the original — the name and all its instrument parameters will be the same. In LOAD mode you can adjust the volume of a copied instrument, select, deselect and stack it as you would any instrument.
- These copied instruments are temporary things — the song or sequence doesn't know that it's playing a copy (in fact the song or sequence never knows what instruments are loaded into which locations). Any copied instruments will be forgotten when the ASR–10 is turned off unless you save the contents of memory as a bank (using the SAVE BANK command on the Command/Instrument page). When you load a bank, the ASR–10 recreates any copied instruments that were there when the bank was saved.
- You cannot edit the instrument, layer, or wavesample parameters of the copy independently. Any parameter changes made to the copy will affect the original, and vice-versa.
- If you load a new instrument into the ***Instrument•Sequence Track*** location that contains the original instrument, any copies of that instrument will be automatically deleted from memory.

MIDI Sequencing on the ASR–10 — MIDI Connections

You can sequence any remote MIDI device exactly as you would a local ASR–10 instrument:

- On the Edit/System•MIDI page, set the TRANSMIT ON parameter to TRANSMIT ON= INST CHAN.
- Create a MIDI instrument as shown in *Section 17 — Sequencing/MIDI Applications* and assign the MIDI instrument to transmit on the MIDI receive channel of the remote MIDI device you want to sequence from this track.
- Now select the MIDI instrument and proceed with recording the track just as you would any other track. All keys, controllers and program changes you record will be transmitted out via MIDI to the remote device. You can mix, mute, and solo these MIDI tracks on the Edit/Track page.

When controlling multiple remote MIDI devices, first connect the various devices to the ASR–10, and to each other, as shown here. Connect the MIDI Out jack of your MIDI Controller to the MIDI In jack of the ASR–10. Connect the MIDI Out jack of the ASR–10 to the MIDI In jack of the first device. Then connect the MIDI Thru jack of the first device to the MIDI In jack of the second device. Connect the MIDI Thru jack of the second device to the MIDI In jack of the third device. And so on, for as many devices as you will be using.



With this arrangement, once you set up the proper MIDI channels, etc. (see below), each remote MIDI device will receive and play only the data that is intended for it, and will "pass along" all other data. Also, each remote MIDI device can be played from its own keyboard without affecting the others, because MIDI Thru jacks only pass along incoming MIDI data, and do not transmit what is played on the instrument.

The above set up is ideal for controlling everything from the ASR–10. Simply by selecting the ***Instrument•Sequence Track*** that is set to transmit on the same MIDI channel as a particular remote MIDI device is set to receive on, you can:

- Play the ASR–10 and other remote MIDI devices from the MIDI keyboard
- Record a track in the ASR–10 sequencer that will play back on the remote MIDI device when you play the sequence or song; and
- Send the remote MIDI devices program changes and adjust their volume (assuming the devices receive MIDI Volume)

Once you have made the appropriate connections, and set up the MIDI configuration of the tracks and all remote MIDI devices, you can use your main keyboard and the ASR–10's front panel to control and record all the MIDI devices in your rig.

MIDI Mode and Channel — Remote MIDI Devices

The next step is to set up each remote MIDI device to receive only the MIDI information that is intended for it. When each remote MIDI device is set to receive on a different MIDI channel (or a number of them, for multi-timbral units) you can control them all right from the ASR–10.

For each remote MIDI device:

- **Set to POLY (OMNI OFF) or MULTI Mode.** Each remote MIDI device must be in a mode where it receives *only* on its selected MIDI channel (or channels). This is usually referred to as POLY (or OMNI OFF) mode for receiving on a single channel, or MULTI mode for receiving independently on multiple channels. Consult the owner's manual if there is any question about a particular device.

- **Select a MIDI receive channel or channels.** The best idea is to assign each remote MIDI device its own MIDI receive channel(s) and leave it that way. If you know, for instance, that a certain synth is always set to receive on MIDI channel 4, you can quickly set up an ***Instrument•Sequence Track*** to drive that synth by simply creating a MIDI instrument (see *Section 17 — Sequencing/MIDI Applications*), then assigning that instrument to transmit on MIDI channel 4. Also when each remote MIDI device is always set to its own distinct MIDI channel, it means that different sequences and songs recorded at different times will always play the right instrument on the right track.

Once you have assigned MIDI receive channels to each MIDI device in your rig, *write them down*, and keep the paper handy for quick reference.

Recording MIDI Tracks

Once everything is set up, you can proceed with recording MIDI instruments exactly as you would for instruments with LOCAL or BOTH status that will play locally. Tracks that are transmitted out via MIDI are treated the same as internal tracks in terms of recording, overdubbing, punching in, editing, etc. Follow the same steps outlined earlier in this section for recording the first track and then for additional tracks.

For each successive track you record, the procedure will follow the same lines:

- Define the MIDI configuration for the ***Instrument•Sequence Track*** ,
- Record the track, and then
- Either keep or discard the new track from the KEEP=OLD/NEW page.

MIDI Instruments can be selected and stacked from the ***Instrument•Sequence Track*** buttons, and can be muted or soloed from the Edit/Track page, the same as any other tracks. Edit/Instrument parameters such as Volume, Key Zone, and Transpose all apply to MIDI tracks just as they do with LOCAL tracks.

Most often you will be recording sequences and songs that play some MIDI instruments and some LOCAL Instruments. When this is the case, be sure that you set the Edit/Instrument MIDI STATUS to LOCAL (as opposed to BOTH) for the instruments on the tracks that you want to play only on the ASR–10. This will avoid accidentally transmitting unintended MIDI data to any external MIDI devices when you play the sequence or song.

Recording Program Changes

The ASR–10 allows program changes to be recorded onto any track of a sequence at any point within the sequence.

To record a program change onto a track:

- Make sure that RECORD MODE=ADD in the Edit/Seq•Song page.
- Enter Overdub mode by pressing ***Play***, followed by the ***Record*** button. The "ODUB" indicator light will begin flashing.
- While holding down the appropriate ***Instrument•Sequence Track*** button (the one that corresponds to the track onto which you wish to record the program change), use the numeric keypad to type in the program number you want to be recorded.
- *Release* the ***Instrument•Sequence Track*** button at the point within the sequence that the program change should occur. The program change will be recorded on the track *the moment the button is released.*

Recording into the ASR–10 Sequencer from a MIDI Controller

On many occasions, you may find it convenient to use an external controller to record a track into the ASR–10's internal sequencer. This could be for added authenticity (recording from Drum Pads to get a "live drummer" feel), performance flexibility (recording from a weighted-action MIDI keyboard controller like the ENSONIQ KS–32), or ease of use (a guitarist recording from a guitar controller). No matter which type of MIDI controller you use, there are a few parameters that need to be set in order to record into the sequencer.

Recording Onto a Single Track (One MIDI Channel)

- Connect a MIDI cable from the MIDI Out of the controller to the MIDI In of the ASR–10.
- Set the MIDI IN MODE to POLY on the Edit/System•MIDI page. The MIDI controller should be in POLY mode as well.
- Set the ASR–10 MIDI BASE CHANNEL to the same MIDI channel that the MIDI controller is set to transmit on.
- Select the track you want to record on by pressing the appropriate ***Instrument•Sequence Track*** button on the ASR–10. Make sure the Instrument on the selected ***Instrument•Sequence Track*** has its Edit/Instrument MIDI STATUS set to BOTH, LOCAL, or MIDI.
- On the Edit/Seq•Song page, set the SEQ REC SOURCE parameter to BOTH or MIDI.

Recording proceeds normally from this point, with the sequencer responding to the data from the MIDI controller the same as it would to data from its own keyboard.

Recording Onto Several Tracks (Multiple MIDI Channels) — Multi-Track Record

When (Edit/Seq•Song) SEQ REC SOURCE=MULTI, the ASR–10 will record incoming MIDI information simultaneously on *all* tracks that have an instrument loaded. This is ideal for downloading a sequence created on an external sequencer all in one pass. The ASR–10 will not record notes played on its own keyboard. Each track should be set to a different MULTI-IN MIDI CHAN on the Edit/Track page. If two tracks have the same MULTI-IN MIDI CHAN, only the lower numbered track will be recorded.

- Connect a MIDI cable from the MIDI Out of the external sequencer to the MIDI In of the ASR–10.
- Set the MIDI IN MODE to either MONO B (for guitar controllers sending on individual strings) or MULTI (for transferring data from another sequencer) on the Edit/System•MIDI page.
- Disable countoff on *both* systems to avoid problems. On the ASR–10, this is found on the Edit/Seq•Song page.
- Disable sequence looping on the external sequencer (LOOP=OFF).
- Create a new sequence with the correct time signature on the ASR–10.

- Set the individual MULTI-IN MIDI CHANNEL (found on the Edit/Track page) for each track, matching MIDI channels with those that will be transmitted on by the external sequencer. Different MIDI receive channels must be selected for each ***Instrument•Sequence Track*** that you want to receive via MIDI. If more than one sequence track is set to the same MULTI IN MIDI CHAN, only the lowest numbered ***Instrument•Sequence Track*** will receive via MIDI.
- Make sure that the Instruments on all of the tracks you want to record on have their Edit/Instrument MIDI STATUS set to BOTH, LOCAL, or MIDI.
- On the Edit/Seq•Song page, set the SEQ REC SOURCE parameter to MULTI.
- If recording from an external sequencer, set the ASR–10 to sync to MIDI clocks. On the Edit/Seq•Song page, select CLOCK SOURCE=MIDI. Set this parameter to INTERNAL if recording "live" from a MIDI guitar or other multi-channel controller.

With these parameters set, pressing the ***Record*** button places the ASR–10 into Record Waiting mode (the display will flash WAITING. . .). If external sync is used (CLOCK SOURCE=MIDI), recording begins with the first MIDI clock received after the MIDI Start command. If internal sync is selected (CLOCK SOURCE=INT), then recording begins when the first note is received.

Once the ***Stop•Continue*** button has been pressed, the ASR–10 displays the ###BARS-KEEP TRACK? message. Pressing the ***Enter•Yes*** button will save the data. The EDITING DATA... message is displayed while the sequencer is processing the data you have recorded. The time required for processing depends on how much data and how many tracks were recorded. It can take several minutes. You can either re-record or erase the tracks (or the entire sequence or song) if you do not wish to keep the results after recording.

Multi-Track Record always operates in REPLACE mode, and when SEQ REC MODE= MULTI the SEQ REC MODE parameter on the Edit/Seq•Song page is automatically set to REPLACE, and cannot be edited. However, while tracks are being rerecorded in MULTI, both the original and new tracks can be heard. This can be confusing, but only the new tracks are being recorded, and when played back the sequence will sound fine.

Note: When SEQ REC SRC= MULTI, the sequencer will always record both channel and key pressure received via MIDI, no matter what the Edit/Instrument PRESSURE MODE parameter is set to.

For more information about sequencing, see *Section 17 — Sequencing/MIDI Applications*.

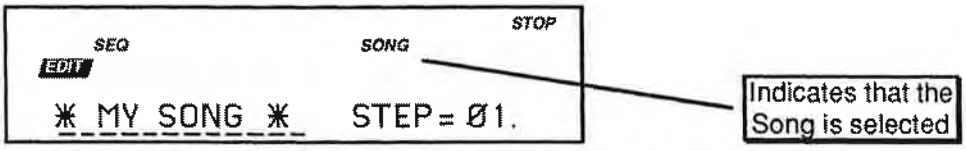
Song Mode

The ASR–10's Song mode allows you to chain individual sequences together to form a song. Since each sequence has a fixed length, tempo, and time signature, you can use the Song mode to create works with tempo and time signature changes. However, Song mode on the ASR–10 lets you do much more than just play sequences in order.

First, for each step of the song you can selectively mute or transpose any of the tracks in the sequence. And second...

You can record in song mode. An ASR–10 song offers you another set of 8 tracks, called song tracks, which let you record over the entire length of the song. The information recorded in these song tracks is separate from information in the tracks of the individual sequences which comprise the song. However, each song track *does* "share" an instrument with the same-numbered sequence track. The following are a few things to remember about ASR–10 song mode:

- There is only one song in the ASR–10 at a time. Whenever the song is selected, (as opposed to one of the sequences) the SONG indicator lights on the display:



The SONG indicator remains lit as long as the song is selected, no matter what mode the ASR–10 is in or page it is on. Whenever the SONG indicator is *not* lit, an individual sequence is selected.

- You select the song in the same way that you select sequences, on the Edit/Seq•Song page (press ***Edit***, then double-click on ***Seq•Song***).

- Songs are composed of steps. For each step of the song, you define:
  - Which sequence will play during that step;
  - The number of repetitions (REP) – how many times the sequence will play during the step;
  - The status (Play, Mute, or Transpose) of each track of the sequence during the step; and
  - The transpose amount, which determines how far any transposed tracks in that step will be shifted up or down in pitch.

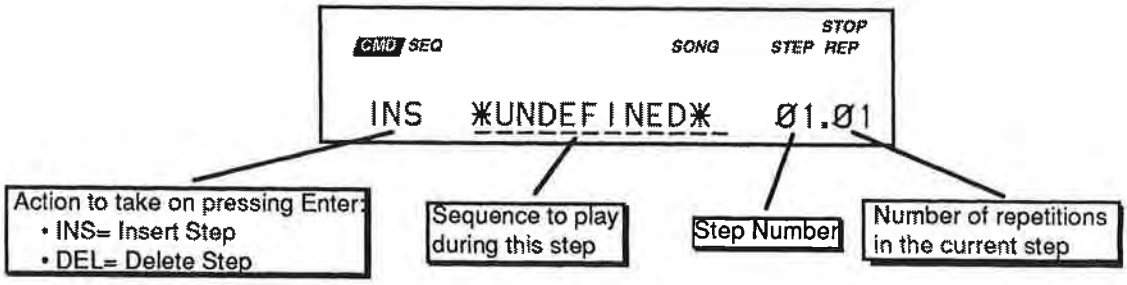
A song can contain as many as 99 steps, with up to 63 repetitions of each step.

- Putting together an ASR–10 song involves the following steps:
  - Record (or load from disk) the various sequences that you want to combine to form the song.
  - Construct the song, using the Edit SONG STEPS command on the Command/Seq•Song page (see below).
  - Record any song tracks.
  - Save the song (along with its component sequences) using the SAVE SONG + ALL SEQS command on the Command/Seq•Song page.
  - As a final step, it's always a good idea to save the contents of memory as a bank, so that you can later load the song and load all the instruments into the proper locations, just by loading the bank.

Making a Song (Command/Seq•Song, EDIT SONG STEPS)

Once you have recorded a number of sequences that you want to chain together into a song:

- Press ***Command***, then ***Seq•Song.***
- Press the ***Left/Right Arrow*** buttons until the display shows EDIT SONG STEPS.
- Press ***Enter•Yes***. The following page appears:



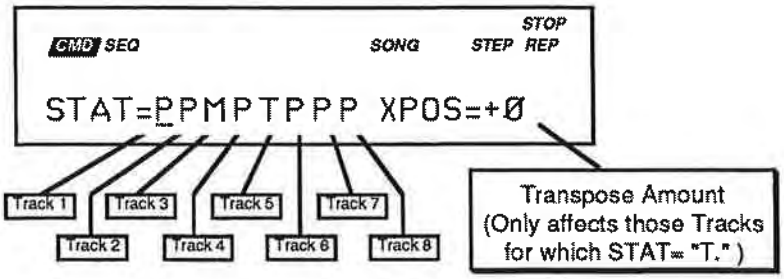
There are four things that can be selected (underlined) on this screen:

- On the left you can choose INS (insert step) or DEL (delete step). This is normally set to INS, for inserting new steps into your song (see the following page for details about deleting steps).
- In the middle of the screen you choose which sequence will play for this song step. This space currently reads *UNDEFINED*, as shown above, when no steps have yet been defined for the song.
- To the right of the display you see the step number (which in the case of a new song will be step 1) and the number of repetitions (or reps) for that step.

For each song step that you want to create:

- Move the cursor (if it's not already there) to underline the sequence name, which currently reads *UNDEFINED*.
- Press the ***Up/Down Arrow*** buttons to select among the sequences in memory until the display is showing the name of the sequence you want to play during that step.
- Scroll right to move the cursor to the number of repetitions (REP), and adjust the number of times you want the sequence to play during the step. (If you only want the sequence to play once during the song step, leave it set to 01.)

- Press the ***Right Arrow*** button again to reveal the following TRACK STATUS display:



Here, for each step of the song, you can select Play, Mute, or Transpose Status for each track of the sequence.

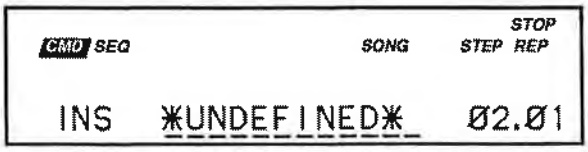
- "P" means PLAY - tracks that show a "P" will play normally.
- "M" means MUTE - selecting "M" for a track will silence that track during the song step.
- "T" means TRANSPOSE - tracks that show a "T" will be transposed up or down, by the amount shown to the right, for the duration of the step.

If you want to mute or transpose any of the tracks during that step, scroll until that track is underlined, and use the ***Up/Down Arrow*** buttons to select "M" (mute) or "T" (transpose) for that track.

If you have selected "T" to transpose any of the tracks during the song step, scroll right to underline the Transpose (XPOS) Amount. This can be adjusted in one-semitone steps from -12 (down one octave) to +12 (up one octave). This control will only affect tracks for which "T" Status has been chosen. Conversely, if you choose "T" status for one or more tracks, but leave the Transpose Amount set at zero, it will have no effect.

- Press the ***Left*** or ***Right Arrow*** button to return to the first Song Step Editor screen.

Once the sequence and number of REPs are correct, press ***Enter•Yes*** to create step 1 of the song. The display will briefly read "EDITING...," and then will display step 2:



Now repeat the above procedure to define step two of the song.

For each step, choose a sequence and a number of repetitions and then press ***Enter•Yes*** to create the step.

- There is always one *UNDEFINED* step *after* the last step in the song.

To Go to a Different Step in the Song:

Move the cursor to underline the step number, and use the ***Up/Down Arrow*** buttons to go to any step within the song. After you have finished editing the song steps (or at any point during the process, for that matter) you can go back through the song to check that all the steps are right.

To Change Anything in an Existing Song Step:

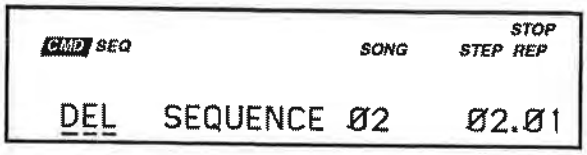
To change any of the variables (sequence to play, number of reps, track mute or transpose) within a song step that has already been created, simply go to that step, as described above, select the variable you want to change and change it. You don't have to press ***Enter•Yes*** to make the changes permanent. You only have to press ***Enter•Yes*** to actually insert or delete a step.

To Insert a Step Anywhere in the Song:

- Go to the step that you want the new step to precede. That is, if you want to insert a step between step 2 and step 3, go to step 3.
- Move the cursor to underline the sequence name.
- Use the ***Up/Down Arrow*** buttons to select the sequence you want to play for the new step.
- Set the number of repetitions and any mute or transpose settings for the step as shown earlier.
- Press ***Enter•Yes***. The new step is inserted. If, as in the example above, you go to step 3 and insert a step, the step you inserted becomes the new step 3. What was step 3 becomes step 4, and so on.

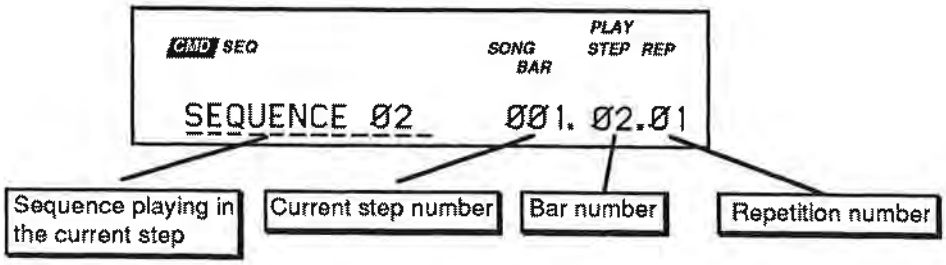
To Delete a Step Anywhere in the Song:

- Go to the step that you want to delete.
- Move the cursor to underline INS at the left of the display.



- Press the ***Up Arrow*** button to change INS (insert) to DEL (delete).
- Press ***Enter•Yes***. The step that was showing will be deleted.

When you are through editing the song, press ***Cancel•No*** to exit the song step editor. Press ***Edit*** and then double-click on ***Seq•Song*** to go the Sequence/Song Select parameter. You will see that the song is selected. Press ***Play*** to play the song. When playing back a song from this page, the display shows:



Song Tracks

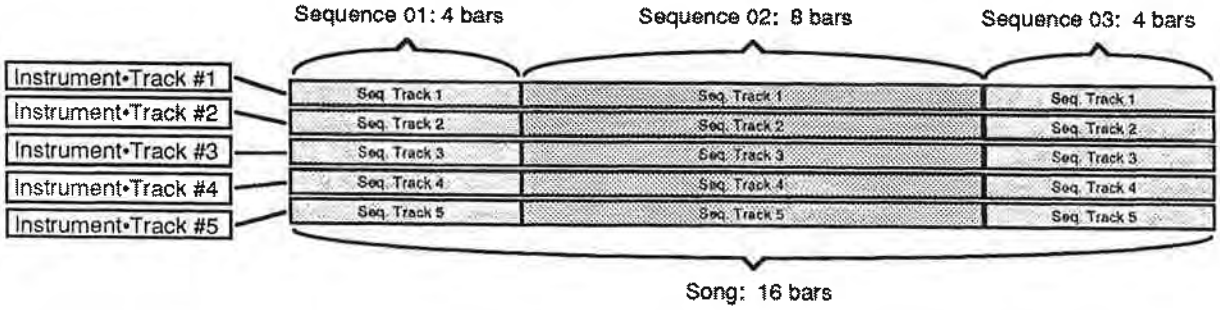
A song on the ASR–10 is much more than simply a group of sequences chained together. Once you have created a song and edited its steps, you can record another complete set of 8 song-length tracks. The length of the song tracks is determined by the combined length of the song's component sequences.

Recording Song Tracks

Let's suppose you have constructed a song. For our example we will take a simple case:

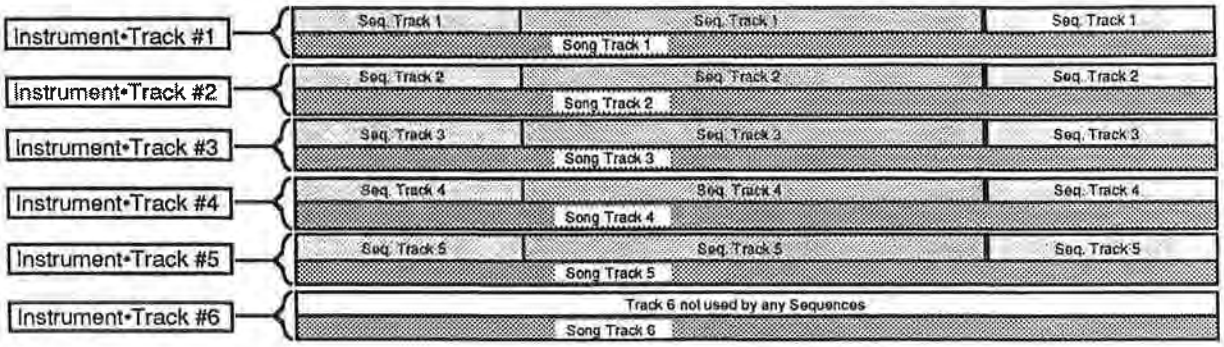
- You have recorded three sequences, each using five tracks, and combined them into a song.
- Step one of the song is Sequence 01 (a 4-bar sequence) for 1 repetition;
- Step two is Sequence 02 (an 8-bar sequence) for 1 repetition; and
- Step three is Sequence 03 (a 4-bar sequence) for 1 repetition.

Your song would look like this:



Now, with the song selected, you can enter Record (by holding down ***Record*** and pressing ***Play***) and record a new song-length track on any of the five tracks. Follow the same procedures for recording song tracks that you would for sequence tracks. The only difference is that a song track is associated with the song itself and not with the individual sequences that comprise the song.

Continuing with the previous example, the song tracks might look like this:



Any notes and controllers recorded in a song track will play the instrument loaded into that location — song tracks "share" the eight ***Instrument•Sequence Track*** locations with the corresponding sequence tracks. Therefore they are not entirely independent. For example pitch bends, sustain pedal, pressure, and any other controllers recorded into a song track will also affect notes recorded on the corresponding sequence tracks.

For this reason it is often a good idea to record a song track in a location not used by any sequences, as with track six in the illustration above. You could load a piano sound into ***Instrument•Sequence Track*** 6 and record a song track with that sound that would remain completely independent of any tracks in the individual sequences. Or, make a copy of an instrument in an unused ***Instrument•Sequence Track*** location and record a song track using the copy.

- When the song is selected, if you go to the Edit/Track page you will find a third screen has been added — it looks just like the track status screen for a sequence except it says SNG instead of SEQ in the left corner of the display. This shows you the status of the song tracks. You can select P (play), M (mute), or S (solo) for any of the song tracks just as would sequence tracks. Note, however, that changing the mix or pan of a song track will also affect any corresponding sequence tracks.

Note: Any recording or track editing performed while in song mode will affect the song tracks, not the tracks of the individual sequences. To do any track editing (or additional recording) within the individual sequences of a song, you must first select that sequence from the Edit/Seq•Song page.

Recording MIXDOWN VOLUME and MIXDOWN PAN

The song tracks can record a dynamic mixdown that allows you to record volume and pan changes over time. You can record *continuous* volume changes, by editing the Edit/Track MIX parameter while recording with SEQ REC MODE= ADD on a song track. This will record MIXDOWN VOLUME events onto the selected song track, overriding the default MIX setting for the corresponding sequence track. This allows you to record fade in and out mixdown information across multiple sequences. In addition, MIXDOWN PAN events can be recorded by editing the Edit/Track PAN parameter while recording with SEQ REC MODE= ADD. Here's how:

1) Select the song by pressing ***Edit***, then ***Seq•Song***. Double-click the ***Seq•Song*** button, and use the ***Data Entry Controls*** to select the song (the SONG indicator will light).
2) Scroll left and set SEQ REC MODE= ADD. This will prevent any unwanted changes to existing song track events.
3) Select the ***Instrument•Sequence Track*** that corresponds to the song track on which you want to record MIXDOWN VOLUME.
4) While holding down ***Record***, press ***Play***.
5) Press ***Edit***, then ***Track***, and when the sequencer enters Record/Overdub mode, adjust the MIX value while the sequencer plays back. The volume changes you hear will be recorded as MIXDOWN VOLUME events.
6) Press the ***Stop•Continue*** button, and press ***Play*** to listen to your newly recorded mix. If your happy with the results, select KEEP= NEW on the Audition page, and then press ***Enter•Yes*** to keep the results. The new MIXDOWN VOLUME events will now be permanently recorded on the song track.

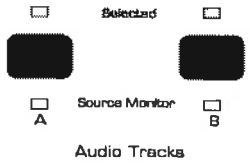
- Follow the same procedure to record MIXDOWN PAN events. At step 5 however, edit the PAN value.

About Audio Tracks

Audio Tracks serve two basic purposes:

- Sing/Play through them — The Audio Tracks allow continuous audio monitoring of the stereo audio inputs through effects (if desired), enabling you to sing along (or play a guitar) through effects while playing a sequence, for example.
- Sample through them — The signal monitored through the Audio Tracks (mono or stereo) is the signal that will be sampled.

The ASR–10 has two Audio Track buttons, labeled ***A*** and ***B***. Each audio track has its own mix, pan, and output bus assignment, as set on the Edit/Track page. The two ***Audio Track*** buttons are located to the left of the ***Instrument•Sequence Track*** buttons:



Each ***Audio Track*** button has 2 LEDs located above and below it:

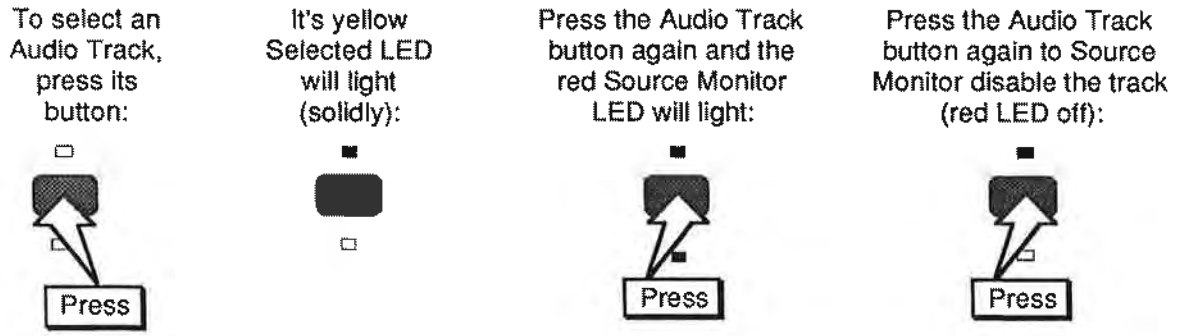
- The lower LED is red, and is labelled "Source Monitor." When solidly lit, it indicates that the Audio Track Record Source (set on the Sample•Source Select/REC SRC page) can be monitored on the Audio Track, and that a voice is being used to monitor this audio signal. When the Source Monitor LED is off, the Audio Track is muted and inaudible.

  When REC SRC= MAIN-OUT, the audio inputs are disabled, and both Source Monitor LEDs remain off at all times. This happens because REC SRC= MAIN-OUT recording is all ASR–10 generated audio that is routed to BUS1, 2, or 3, and the Source Monitor voices are not needed to monitor this REC SRC, since it is always audible out the Main Outs.

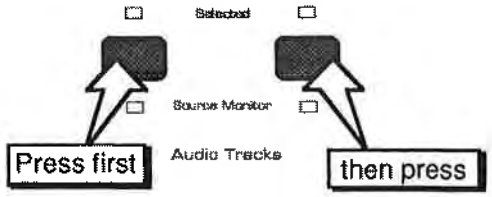
- The upper LED is yellow, and is labelled "Selected." When solidly lit, it indicates that the track is selected for parameter editing, and that the Audio Signal Source being monitored on the track is selected for sampling. When the "Selected" LED is off, the track's Edit/(audio) Track parameters cannot be edited, and the Audio Signal Source being monitored on the track will not be sampled.

Selecting, Enabling, and Disabling Audio Tracks

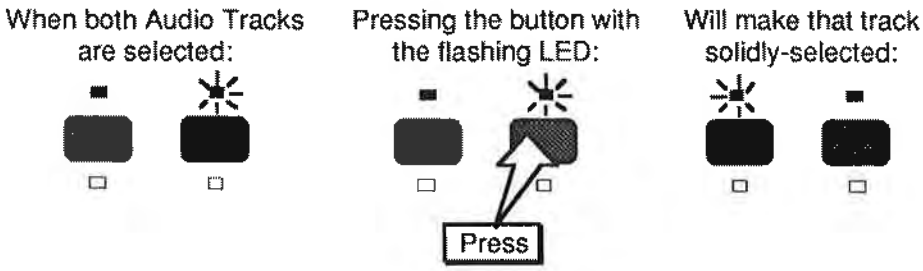
When you want to select a single Audio Track that is not already selected, press its ***Audio Track*** button. It's yellow LED will light. When an Audio Track is selected (yellow LED lit), successive presses of the ***Audio Track*** button will alternate between Source Monitor enable (red LED on) and Source Monitor disable (red LED off) for that track.


To select an Audio Track, press its button:
It's yellow Selected LED will light (solidly):
Press the Audio Track button again and the red Source Monitor LED will light:
Press the Audio Track button again to Source Monitor disable the track (red LED off):
Press
Press
Press

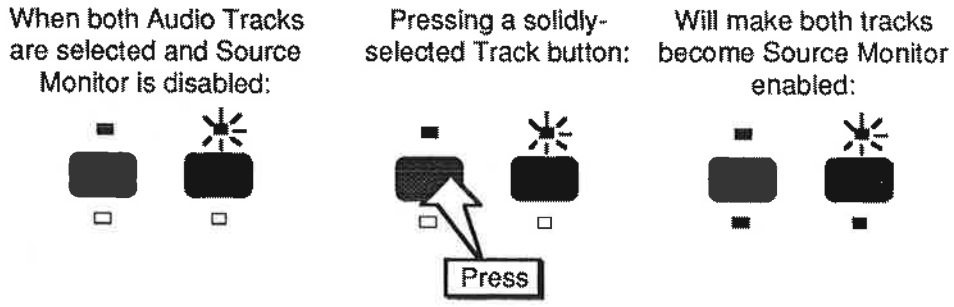
When you want to select both Audio Tracks, hold down one ***Audio Track*** button, and press the other ***Audio Track*** button. Both Audio Tracks will become Selected (both yellow LEDs lit) and Source Monitor enabled (both red LEDs lit).



When both Audio Tracks are selected (both yellow LEDs lit), pressing the Audio Track button with the flashing yellow LED will cause that track to become solidly-selected (solidly lit). The previous solidly-selected Audio Tracks will become flashing-selected.


When both Audio Tracks are selected:
Pressing the button with the flashing LED:
Will make that track solidly-selected:
Press

When both Audio Tracks are selected (both yellow LEDs lit) and Source Monitor disabled (both red LEDs off), pressing a solidly-selected Audio Track button will cause both Audio Tracks to become Source Monitor enabled (both red LEDs on). Successive presses of the solidly-selected Audio Track will alternate between Source Monitor disable and Source Monitor enable for both tracks.


When both Audio Tracks are selected and Source Monitor is disabled:
Pressing a solidly-selected Track button:
Will make both tracks become Source Monitor enabled:
Press

- When both Audio Tracks are selected (both yellow LEDs lit), they will both remain selected in Edit and Command Modes. The solidly-selected Audio Track is considered selected for parameter editing in Edit Mode, and as the target for track commands in Command Mode. In Edit and Command Modes, both Audio Tracks will be deselected as a pair (both Audio Track yellow LEDs off), when an ***Instrument•Sequence Track*** is selected.

Note: Audio input being monitored on an audio track will be muted during certain digital processing operations.

Sequencer & Audio Track Parameters

Section 13 — Sequencer and Audio Track Parameters

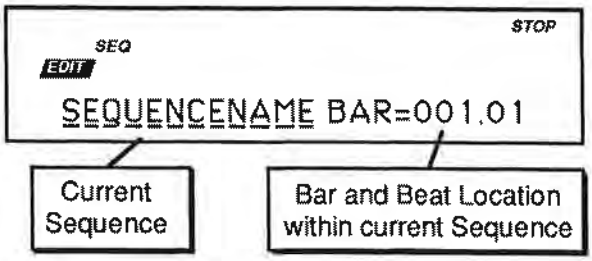
The following parameters control the ASR–10 sequencer. Parameters for creating and manipulating sequences, the song, and the tracks contained within each are covered here. This section also describes the Audio Track Edit/Track parameters. For a basic overview of the concepts involved, refer to the previous section.

The Edit/Seq•Song Page

This page acts as the main "control panel" for sequencing.

EDIT SEQ•SONG	CURRENT SEQUENCE/SONG - BAR - BEAT/STEP
	Press Edit / Seq•Song / 0 (for Current Seq/Song) 1 (for Bar)

Current Sequence/Song



This screen is where you select a sequence or the song. With the selected sequence or song name underlined (as shown above), use the ***Up/Down Arrow*** buttons or the ***Data Entry Slider*** to select a different sequence or song.

Hint: You can always get back to the Current Sequence/Song screen by double-clicking the ***Seq•Song*** button (when in Edit mode).

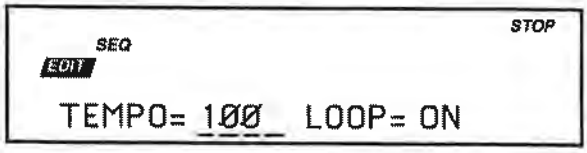
BAR/STEP — The GOTO Function

- Press the ***Right Arrow*** button to move the cursor to the BAR parameter. The display reads as follows:

This parameter shows the current location in the selected sequence or song. When the cursor is under the Bar, Beat (or Step) Number, you can use the GOTO function to "go to" any location within the sequence or song, by moving the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to select a new location, then pressing ***Enter•Yes***. Pressing ***Play*** will *always* start the sequencer from the beginning of the selected sequence. Pressing the ***Stop•Continue*** button will begin playing the sequence from the selected bar.

Hint: After using the GOTO function to go to a certain Bar, Beat (or Step) location, pressing ***Enter•Yes*** twice with the cursor under the Bar, Beat (or Step) value will relocate the sequencer to the last selected GOTO location.

EDIT SEQ•SONG	TEMPO - LOOP
	Press Edit / Seq•Song / 2 (for Tempo) 3 (for Loop)



Here you can adjust the tempo of the sequence in beats per minute, and the LOOP parameter. When LOOP=ON the sequence will repeat continuously, when LOOP=OFF the sequence will play once and stop. When CLOCK SOURCE= MIDI, the TEMPO value will read EXT, to indicate that the tempo is controlled by incoming MIDI clocks.

Range: 20 to 250, EXT

Tap Tempo

When the TEMPO parameter is selected, tapping on the ***Enter•Yes*** button at the correct rate for the desired tempo will update the TEMPO parameter to the new value.

EDIT SEQ•SONG	CLOCK SOURCE
	Press Edit / Seq•Song / 4

This parameter determines the sequencer CLOCK SOURCE. There are two possible choices:

- INTERNAL — The ASR–10 will use its own internal clock.
- MIDI — The ASR–10 will sync to incoming MIDI clocks from a remote device. This is useful when syncing to a drum machine or external sequencer, but still recording on the ASR–10.

When CLOCK SOURCE= MIDI, pressing the ***Record*** button puts the ASR–10 into standby MIDI sync record. The MIDI indicator light will flash and the display will show WAITING... Recording will begin with the first MIDI clock received after a MIDI Start command. If the ***Stop•Continue*** button is pressed before a MIDI Start command is received, the ASR–10 waits 5 seconds before stopping.

EDIT SEQ•SONG	CLICK - CLICK NOTE VALUE
	Press Edit / Seq•Song / 5

These two parameters control the metronome CLICK. The first parameter controls when the click will be heard:

- REC — the click will be heard only when the sequencer is recording, and not when it is playing. This is the default.
- OFF — the click will not be heard during Play or Record.
- ON — the click will be heard whenever the sequencer is running.

The second parameters controls the note value that the click will play:

Range: 1/2, 1/2T, 1/4, 1/4T, 1/8, 1/8T, 1/16, 1/16T, 1/32, 1/32T ("T"= triplets)

EDIT SEQ•SONG	CLICK VOLUME
	Press Edit / Seq•Song / 6

The volume of the click track is adjustable from 0 to 99.

EDIT SEQ•SONG	CLICK PAN - OUT (Output Routing)
	Press Edit / Seq•Song / 7

Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to pan the click track within the stereo field (-99 to +99). The OUT value will route the click to one of the three stereo effect busses (BUS 1, 2, or 3), or send it to one of the three pairs of auxiliary outputs (AUX 1, 2, or 3). The panning and output routing of the click is handled in the same manner as WaveSample panning and output routing.

EDIT SEQ•SONG	SEQ COUNTOFF
	Press Edit / Seq•Song / 8

When:

- SEQ COUNTOFF=OFF — the song or sequence will start playing or recording as soon as you press ***Play***, or ***Record*** and ***Play*** (i.e. there is no count off).
- SEQ COUNTOFF=ON — you will hear one bar of click track before the sequencer goes into Play or Record.
- SEQ COUNTOFF=RECORD — you will hear one bar of click track before the sequencer goes into Record, but not prior to going into Play.
- SEQ COUNTOFF=QUIET — there is a *silent* one-bar count before the sequencer goes into Play or Record.

EDIT SEQ•SONG	SEQ REC MODE
	Press Edit / Seq•Song / 9

The SEQ REC MODE parameter determines how sequence data will be recorded by the ASR–10 sequencer when you enter Record. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to choose between the following options:

- SEQ REC MODE=REPLACE — Anything recorded into an existing track will replace data that was in the track previously. The sequencer will stop recording after reaching the end of the sequence or song, and will enter Audition/Play mode. When the SEQ REC SOURCE= MULTI, REPLACE is automatically selected, and cannot be edited.
- SEQ REC MODE=ADD — New data recorded into an existing track will be added to (or merged with) data already in the track. The existing data will be left intact. Again, the sequencer will exit Record after reaching the end of the sequence or song, and will enter Audition/Play.
- SEQ REC MODE=LOOPED — As with ADD mode, new data recorded into an existing track will be added to (or merged with) data already in the track. However, when SEQ REC MODE=LOOPED, the sequencer will remain in Record for as many times as you play through the song or sequence (rather than dropping out of Record after reaching the end). Think of this as "Drum Machine mode" — as long as the sequence or song keeps playing, you can keep adding parts until you stop the sequencer.

EDIT SEQ•SONG	**SEQ REC SOURCE**
	Press Edit / Seq•Song / scroll using the arrow buttons

This parameter determines what will be recorded when the MIDI IN MODE= MULTI on the Edit/System•MIDI page — data from the keyboard, from incoming MIDI, or both. Setting this parameter to MIDI is useful if you want to send data from the ASR–10 to a MIDI processor, then record only the processed data coming back from the processor. This parameter only has an effect in MULTI mode; in POLY or OMNI modes the sequencer always records both.

- BOTH — The sequencer will record data received either from the keyboard or via MIDI on the track's MULTI IN MIDI channel.
- KYBD — The sequencer will record data only from the keyboard. This setting will only work with the ASR–10 keyboard version, and will not function with the ASR–10 module (which has no keyboard).
- MIDI — The sequencer will record only data received via MIDI on the track's MULTI IN MIDI channel.
- MULTI — This setting allows you to record data onto multiple ASR–10 tracks simultaneously, but only via MIDI. The ASR–10 will not record notes played on its own keyboard. This is most useful for recording multiple tracks sent from external sequencers or played from guitar controllers. The ASR–10 must be set to receive data from MIDI in MULTI mode (MIDI IN MODE=MULTI on the Edit/System•MIDI page) in order for this feature to work correctly. Multi-Track Record always operates in REPLACE mode, and when SEQ REC SOURCE= MULTI, the SEQ REC MODE parameter is automatically set to REPLACE, and cannot be edited.

Command/Seq•Song Page

These commands are used to create, copy, delete, name, and save sequences and the song. Remember, pressing ***Cancel•No*** will cancel these commands.

CMD SEQ•SONG	CREATE NEW SEQUENCE
	Press Command / Seq•Song / 0

Use this command to create a new sequence.
- Select CREATE NEW SEQUENCE.
- Press ***Enter•Yes***. This display allows you to name the sequence, using the Data Entry Controls.
- Press ***Enter•Yes*** and use the ***Left/Right Arrows*** to define the time signature.
- Press ***Enter•Yes*** to complete the command.

CMD SEQ•SONG	COPY SEQUENCE
	Press Command / Seq•Song / 1

This command allows you to copy an entire sequence, including all tracks and sequence parameters. COPY SEQUENCE lets you perform commands and edits on the "copy" while preserving the source sequence.

- Select COPY SEQUENCE.
- Press ***Enter•Yes***. The display reads FROM SEQ=SEQUENCE ##. Choose your source sequence.
- Press ***Enter•Yes***. The display reads NEW NAME=SEQUENCE ##. You can use the data entry controls to change the name of the sequence, or use the default designation.
- Press ***Enter•Yes*** to complete the command.

CMD SEQ•SONG	DELETE SEQUENCE
	Press Command / Seq•Song / 2

Use this command to erase unwanted sequences.

- Select DELETE SEQUENCE.
- Press ***Enter•Yes***. The display reads SEQ=SEQUENCE ##.
- Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to select the sequence you want to delete.
- When the sequence you want to delete is displayed, press ***Enter•Yes***.

CMD SEQ•SONG	SAVE CURRENT SEQUENCE
	Press Command / Seq•Song / 3

Use this command to save the current sequence to disk.

- Make sure the sequence you want to save is selected.
- Select SAVE CURRENT SEQUENCE.
- Press ***Enter•Yes***. The display reads NEW NAME=SEQUENCE ##. If you want to rename your sequence, do so at this time using the data entry controls.
- Press ***Enter•Yes***. The display reads SAVING SEQUENCE ##, and then DISK COMMAND COMPLETED.

CMD SEQ•SONG	SAVE SONG AND ALL SEQS
	Press Command / Seq•Song / 4

Use this command to save all sequencer memory to disk as a file.

- Select SAVE SONG + ALL SEQS.
- Press ***Enter•Yes***. The display reads NEW NAME=–**SONG**. You can change the song name at this point.
- Press ***Enter•Yes***. The display reads SAVING–**SONG**, and then DISK COMMAND COMPLETED.

CMD SEQ•SONG	RENAME SONG/SEQUENCE
	Press Command / Seq•Song / 5

Use this command to rename the current sequence or song.

- Make sure the song or the sequence you want to rename is selected.
- Select RENAME SONG/SEQUENCE.
- Press ***Enter•Yes***. The display reads NEW NAME=SEQUENCE ##. Use the data entry controls to rename the current sequence.
- Press ***Enter•Yes*** to complete the command.

CMD SEQ•SONG	SEQUENCER INFORMATION
	Press Command / Seq•Song / 6

You can use this command to find out the length of the sequence (in bars), the time signature, the size of all sequencer data, and the size of the current sequence. In addition, you can keep track of the duration of the current sequence in minutes, seconds, and fractions of a second. Scroll using the ***Left/Right Arrow*** buttons to view these values. If the song is currently selected, an additional parameter will be displayed: the MIDI SONG SELECT=##.

Hint: When the song is selected, MIDI Song Select numbers can be assigned as follows:
- Select SEQUENCER INFORMATION.
- Press ***Enter•Yes***. Scroll left until the display reads MIDI SONG SELECT=##.
- Select a MIDI Song Select number 0 through 127 with the Data Entry Controls.

The song can be assigned a MIDI Song Select number from 0 through 127. MIDI Song Select numbers 0 through 127 are transmitted via MIDI when you load a new song into the ASR–10, or when the currently loaded song is selected. MIDI Song Selects can be used to instruct a remote sequencer or drum machine to select a new song. The MIDI SONG SELECT parameter (on the Edit/System•MIDI page) must be set to "ON" for this to work. MIDI Song Select numbers are stored with the song when you save the song to disk.

Note: This page is mainly for information *only*. You cannot change any of the parameters (except for the MIDI SONG SELECT number) from this page.

CMD SEQ•SONG	ERASE SONG + ALL SEQS
	Press Command / Seq•Song / 7

Using this command clears all the sequencer memory. Save to disk all sequences that are important to you before performing this command.

- Select ERASE SONG + ALL SEQS.
- Press ***Enter•Yes***. The display reads ERASE ALL SEQ DATA?
- Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to abort the procedure).

CMD SEQ•SONG	APPEND SEQUENCE
	Press Command / Seq•Song / 8

This command allows a sequence to be attached (appended) to the end of another sequence. The FROM SEQ= (source) will be attached to the end of the TO SEQ= (destination). You can also append a sequence to itself to make a longer sequence. For example, if you have a drum sequence that is four bars long, you could append it to itself to create an eight bar sequence. Do it again and you have a sixteen bar sequence. The source sequence must have the same time signature as the destination sequence.

- Select APPEND SEQUENCE.
- Press ***Enter•Yes***. The display reads FROM SEQ=SEQUENCE ##. This is the source sequence. Use the ***Up/Down Arrow*** buttons to select the source sequence.
- Press ***Enter•Yes***. The display reads TO SEQ=SEQUENCE ##. This is the destination sequence. Use the ***Up/Down Arrow*** buttons to select the destination sequence.
- Press ***Enter•Yes*** to complete the command. The source sequence will be attached to the end of the destination sequence. Press ***Cancel•No*** to abort the procedure.

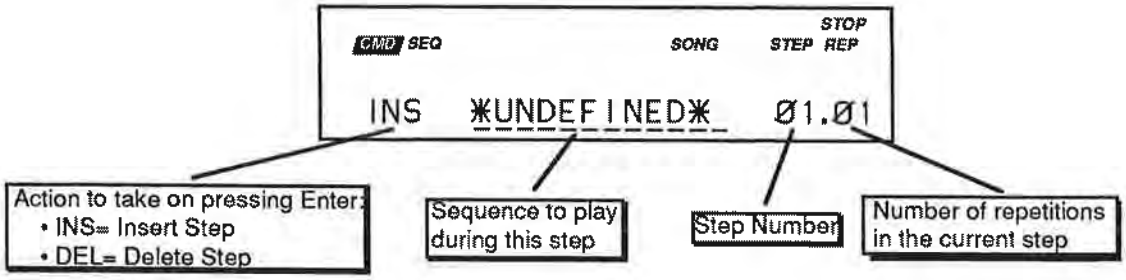
CMD SEQ•SONG	CHANGE SEQUENCE LENGTH
	Press Command / Seq•Song / 9

Use this command to add bars to or delete bars from a sequence.

- Select CHANGE SEQUENCE LENGTH.
- Press ***Enter•Yes***. The display reads ADD ## BARS AT ##.
  If, for example, you wanted to add 2 bars to your sequence starting at bar 4, adjust the display to look like this: ADD 2 BARS AT 4.
- Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to abort the procedure).
- Or, you can underline ADD in the display and use the ***Up Arrow*** button to change it to DELETE. You can now delete specific bars from the sequence using the above procedure.
- Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to abort the procedure).

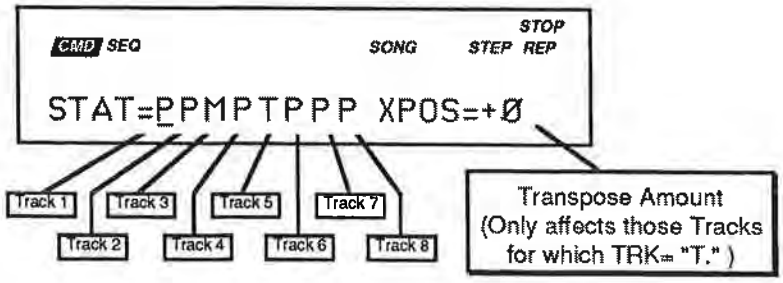
CMD SEQ•SONG	EDIT SONG STEPS
	Press Command / Seq•Song / scroll using the arrow buttons

This command is used to link sequences together to create a song. To do this:
- Select EDIT SONG STEPS.
- Press ***Enter•Yes***. The display shows:



There are four things that can be selected (underlined) on this screen:
- On the left you can choose INS (insert step) or DEL (delete step). This is normally set to INS, for inserting new steps into your song (use the ***Up/Down Arrow*** buttons and select DEL to delete the step).
- In the middle of the screen you choose which sequence will play for this song step. This space currently reads *UNDEFINED*, as shown above, when no steps have yet been defined for the song.
- To the right of the display you see the step number (which in the case of a new song will be step 01) and the number of repetitions (or reps) for that step.

- Press the ***Right Arrow*** button until you see the following TRACK STATUS display:



Here, for each step of the song, you can select Play, Mute, or Transpose Status for each track of the sequence.
- "P" means PLAY — tracks that show a "P" will play normally.
- "M" means MUTE — selecting "M" for a track will silence that track during the song step.
- "T" means TRANSPOSE — tracks that show a "T" will be transposed up or down, by the amount shown to the right (in whole steps), for the duration of the step.

See *Section 12 — Sequencer and Audio Track Concepts* for a more detailed description of EDIT SONG STEPS.

Edit/(sequence) Track Page

These parameters control the sequencer tracks. The parameter values displayed apply to the currently selected sequence track (LED solidly lit yellow). The settings for MIX, PAN, OUT, and EFFECT MOD CONTROL are saved with the bank. To select a track for editing, press the desired ***Instrument•Sequence Track***.

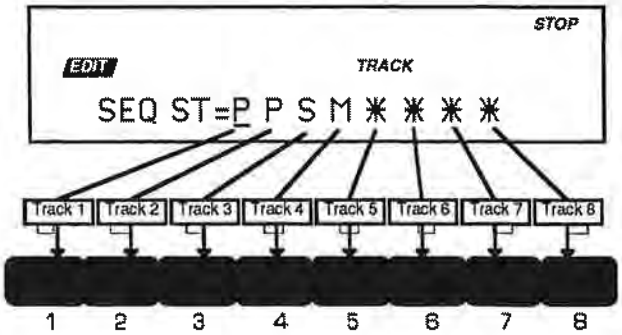
Hint: When Edit/Track parameters are displayed, pressing a loaded ***Instrument•Sequence Track*** button will display the Edit/Track values for the selected track. This feature provides quick access to the same parameter for different tracks when setting track parameter values relative to one another.

EDIT TRACK	**SEQ TRACK STATUS (Mute/Play/Solo)**
	Press Edit / Track / 0

EDIT TRACK	**SNG TRACK STATUS (Mute/Play/Solo)**
	Press Edit / Track / 3 (when a song is selected)

The track status parameter allows you to control the playback status of each track within the selected song or sequence. The cursor (underline) tells you which track is selected. To select a different track, press its ***Instrument•Sequence Track*** button. The underline will move to that track.

You can select any of the tracks which contain recorded data and use the ***Up/Down Arrow*** buttons to select PLAY, SOLO, or MUTE status.



The display shows the status of each track:

- "P" means PLAY — Tracks that show a "P" will play normally.
- "S" means SOLO — Selecting "S" for any track will "solo" the track muting all the others.
- "M" means MUTE — Selecting "M" for a track will silence that track.
- A "Star" indicates that there is nothing recorded on that track (no track data). In the illustration, Tracks 5, 6, 7, and 8 contain no sequence data.

When the song is selected, if you go to the Edit/Track page you will find an additional parameter screen has been added — it looks just like the track status screen for a sequence except it says SNG instead of SEQ in the left corner of the display. This shows you the status of the song tracks. You can select "P" (play), "M" (mute), or "S" (solo) for any of the song tracks just as you would for sequence tracks.

EDIT TRACK	MIX - PAN
	Press Edit / Track / 1 (for Mix) 2 (for Pan)

Track MIX

The MIX parameter controls the volume for each ***Instrument•Sequence Track***. The ASR–10 stores a default MIX setting for each track of a sequence. When you record/overdub a sequence track, the default MIX setting will be recorded onto the track. When you play a sequence back, each track will have the correct volume balance.

Range: 0 to 99

Hint: When TRANSMIT ON = INST CHAN, on the Edit/System•MIDI page, editing the MIX value will transmit MIDI Volume messages (Controller #7) on the selected ***Instrument•Sequence Track's*** MIDI OUT CHANNEL (on the Edit/Instrument page).

Recording the Default Track MIX Setting

Each ***Instrument•Sequence Track*** has its own default MIX setting that is recorded each time that a sequence track is recorded/overdubbed. This default MIX setting is fixed, and applies to the entire duration of the track (unlike *continuous* volume changes recorded with the Foot Pedal, or MIXDOWN VOLUME events recorded on the songs tracks — as described in the previous Section). If the Edit/Track MIX value is edited during playback, the default track MIX setting will be restored each time the sequence loops around. To permanently change the default MIX setting for a track:

- Set SEQ REC MODE= ADD on the Edit/Seq•Song page. This will prevent any unwanted changes to existing track events.
- Select the ***Instrument•Sequence Track*** for which you want to adjust the MIX.
- Press ***Play***, and then press ***Edit***, followed by ***Track***, and adjust the MIX value while listening to the track until you find the proper level.
- Once you've decided on the MIX value, press ***Stop•Continue***.
- While holding down ***Record***, press ***Play***. After the first bar of the sequence has been recorded, press ***Stop•Continue***.
- The KEEP= OLD/NEW Audition page will be displayed. You can't audition the volume change on the KEEP=OLD/NEW page. Select KEEP=NEW and press ***Enter•Yes*** to keep the results. Press ***Play***. You will now hear the new default MIX setting, and it will now be permanently recorded on the track. If you don't like the new default MIX setting, simply repeat the above procedure and set a new MIX level.
- It's a good idea to return to the Edit/Seq•Song page to set the SEQ REC MODE back to REPLACE for later sequencing.

Hint: You can use the optional CVP-1 Foot Pedal to record *continuous* volume changes onto a sequence track. Set PEDAL=VOLUME on the Edit/System•MIDI page, and then record the Pedal•CV controlled volume changes with SEQ REC MODE= ADD. This will record volume (MIDI Controller #7) events onto the track, which can then be edited like any other sequencer event.

For information about mixing down song tracks, see **Recording MIXDOWN VOLUME and MIXDOWN PAN**, in the previous Section.

Track PAN

The PAN parameter controls the current pan position for each ***Instrument•Sequence Track***. This parameter allows you to give each track in a sequence or the song its own specific PAN setting, overriding the WaveSample PAN settings (on the Edit/Amp page).

Range: -99 (full left) to +99 (full right), or WAVESAMPLE
When WAVESAMPLE is selected, each ***Instrument•Sequence Track*** will use the Edit/Amp PAN settings that are contained within the instrument.

EDIT TRACK	**OUT (Output Routing)**
	Press Edit / Track / 4

The OUT parameter controls the output routing for each ***Instrument•Sequence Track***. This parameter allows you to override the Edit/Amp OUT settings for an instrument, and reroute local sequence/song track playback to a specific output bus.

Range: WAVESAMPLE, BUS1, BUS2, BUS3, AUX1, AUX2, or AUX3

When WAVESAMPLE is selected, each ***Instrument•Sequence Track*** will use the Edit/Amp OUT settings that are contained within the Instrument.

Note: When Sample•Source Select/REC SRC Field 1 is set to INPUT+FX, all Voices go DRY, and any ***Instrument•Sequence Tracks*** that were routed to BUS1, 2, or 3 will all show OUT= DRY - - FX BUS IN USE on this screen.

Note: Changing the MIX, PAN or OUT (Output Routing) of a song track will also affect any corresponding sequence tracks.

EDIT TRACK	**EFFECT MOD CONTROL**
	Press Edit / Track / 5

Because of the dynamic nature of the ASR–10's signal processor, instruments can interact and control certain parameters of the selected effect. But what happens when you are using a BANK or ROM effect, one that is accessed globally by all the instruments in memory? Without a way to determine which of the instruments will have control of the effect, controllers from all of the instruments would send confusing messages to the signal processor.

This parameter is specifically designed to be used when a sequence or song is using a BANK or ROM effect, and acts as a "traffic cop" for the instrument controllers.

- ON — The ***Instrument•Sequence Track*** will control the effect with its controllers. Any parameters within the effect that respond to controllers will take their cues from this ***Instrument•Sequence Track***.
- OFF — Controller information generated by the ***Instrument•Sequence Track*** will be ignored by the signal processor.

Obviously, since this parameter is designed to let one instrument control the effect, you will want to set one ***Instrument•Sequence Track*** to EFFECT MOD CONTROL= ON, and all others to OFF.

EDIT TRACK	MULTI-IN MIDI CHAN
	Press Edit / Track / 6

Determines which MIDI channel (1 to 16) the selected ***Instrument•Sequence Track*** will receive on if the MIDI IN MODE=MULTI or MONO B on the Edit/System•MIDI page. Remember, the Edit/Instrument MIDI STATUS parameter has to be set to LOCAL, BOTH, or MIDI to receive incoming MIDI data.

This parameter has no bearing on which channel the ***Instrument•Sequence Track*** will transmit on (that is determined by the MIDI OUT CHANNEL parameter on the Edit/Instrument page).

Note: In MULTI and MONO B modes, different MIDI receive channels must be selected for each ***Instrument•Sequence Track*** that you want to receive via MIDI. If more than one sequence track is set to the same MULTI IN MIDI CHAN, only the lowest numbered ***Instrument•Sequence Track*** will receive via MIDI.

The MULTI-IN MIDI CHANNEL set-up for all eight ***Instrument•Sequence Tracks*** is saved with the global parameters. Therefore, it is a good idea to use the SAVE GLOBAL PARAMETERS command (on the Command/System•MIDI page) to put the MULTI-IN MIDI CHANNEL set-up onto the disk containing your O.S. When the ASR–10 boots up, it will automatically install the MULTI-IN MIDI CHANNEL settings for you.

Hint: If you use several different MIDI reception configurations in your studio, make several copies of the O.S. disk, and save different GLOBAL PARAMETERS on each one. You can then use the LOAD GLOBAL PARAMETERS command (Command/System•MIDI page) to load the different configurations for each O.S. disk.

Edit/(audio) Track Page

These parameters affect the REC SRC Signal Monitor. The parameter values displayed apply to the currently selected audio track (LED solidly lit yellow). To select a track for editing, press the desired ***Audio Track*** button.

The following Edit/(audio) Track parameters are located at the same screen address as the corresponding Edit/(seq) Track screen, and they show the same current parameter.

Pressing an ***Audio Track*** button when the display is showing an Edit/(seq) Track parameter screen that is not applicable to audio tracks, will display the last selected Edit/(audio) Track screen. When the display is showing an Edit/(seq) Track parameter screen that is applicable to audio tracks, pressing an ***Audio Track*** button will display the selected audio track's value(s) for the current parameter. While an audio track is selected, only applicable Edit parameter screens will be subsequently displayed.

- When the Sample•Source Select REC SRC parameter is set to INPUTDRY, the audio tracks monitor the Audio Inputs. The left Audio Input is monitored on ***Audio Track A*** and the right Audio Input on ***Audio Track B***.
- When the Sample•Source Select REC SRC parameter is set to INPUT+FX, the ESP chip is dedicated to processing the Audio Inputs, and the audio tracks monitor the direct output of the ESP chip. The left effects channel is monitored on ***Audio Track A*** and the right channel on ***Audio Track B***. The Wet/Dry mix is set on the Edit/Effects page.
- When the Sample•Source Select REC SRC parameter is set to MAIN-OUT, no audio signals are monitored through the Edit/(audio) Track settings. All other ASR–10 instruments retain their Edit/(seq) Track settings.
- When the Sample•Source Select REC SRC parameter is set to DIGITAL, the audio tracks monitor the Digital Input. The left channel is monitored on ***Audio Track A*** and the right channel on ***Audio Track B***.
- Edit/(audio) Track parameter settings are saved with the global parameters (using the Command/System•MIDI SAVE GLOBAL PARAMETERS command).

EDIT TRACK	**ATRK MIX - PAN**
	Press Edit / Track / 1 (for Mix) 2 (for Pan)

Audio Track MIX

The MIX parameter controls the volume for the signal being monitored on each audio track.

Range: 0 to 99
The default MIX value is 60, because BOOST is permanently ON on the Audio Tracks.

Audio Track PAN

The PAN parameter controls the current pan position for the signal being monitored on each audio track.

Range: -99 (full left) to +99 (full right)
The default PAN settings are ATRK-A PAN= -99, and ATRK-B PAN= +99.

EDIT TRACK	**OUT (Audio Track Output Routing)**
	Press Edit / Track / 4

This parameter allows you to route the signal being monitored on each audio track to a specific output bus.

Range: BUS1, BUS2, BUS3, AUX1, AUX2, or AUX3

Note: When Sample•Source Select/REC SRC Field 1 is set to INPUT+FX, all voices go DRY, and any audio tracks that were routed to BUS1, 2, or 3 will all show OUT= DRY - - FX BUS IN USE on this screen.

Command/Track Page

These commands allow you to manipulate sequence data on the individual tracks.

You can use the ***Data Entry Controls***, or press the keys on the MIDI keyboard connected to the ASR–10 to set the ranges for the following commands.

CMD TRACK	**QUANTIZE TRACK**
	Press Command / Track / 0

This command automatically corrects the timing of a track. After you have recorded a track, the ASR–10 can post-quantize the track and move the notes to the nearest beat boundary. For example, if you set quantize to 1/4, all the notes will be adjusted to the nearest quarter note. You can compare the quantized track with the original track on the Audition page, before deciding whether or not to keep the results.

- Select QUANTIZE TRACK.
- Press ***Enter•Yes***. The display reads TRACK=##. Select the track you want to quantize by pressing its ***Instrument•Sequence Track*** button.
- Press ***Enter•Yes***. The display reads QUANTIZE TO ##### NOTE. The quantize values are: 1/4, 1/4T (triplets), 1/8, 1/8T, 1/16, 1/16T, 1/32, 1/32T, 1/64, 1/64T notes. Select a quantize value.
- Press ***Enter•Yes***. The display reads ENTIRE TRACK. Pressing ***Enter•Yes*** at this point will edit the entire track and place you on the Audition (KEEP=OLD/NEW) page. If you wish to edit only a portion of the track, press the ***Up Arrow*** button until the display reads SET RANGES. You can then press ***Enter•Yes*** and set both the key range and bar/beat range in which the editing will take place. After setting these ranges, the editing will take place and you will be placed on the Audition page. If you've previously set editing ranges for the current song or sequence, you can use the ***Up Arrow*** button to call up USE RANGES. Pressing ***Enter•Yes*** will commence editing based on the previously set range values.
- Press ***Enter•Yes*** if you want to keep the quantized version of the track, or ***Cancel•No*** to keep the unquantized track.

CMD TRACK	COPY TRACK
	Press Command / Track / 1

This command copies the currently selected track (or any bar range of the track) to any other track in any other sequence.

- Select COPY TRACK.
- Press ***Enter•Yes***. The display reads FROM TRACK= ##. Select the source track by pressing its ***Instrument•Sequence Track*** button.
- Press ***Enter•Yes***, or scroll right to select TO SEQ= (sequence name). Use the ***Up/Down Arrow*** buttons to select a destination sequence.
- Scroll right to select TO TRACK= ##. This will default to the lowest numbered empty track. Choose a destination track by pressing one of the active ***Instrument•Sequence Track*** buttons.
- Scroll right to select AT BAR= ##.
- Scroll right to select RANGE= ##. The range parameter lets you select either the entire range, or set your own ranges; allowing you to define the key range, and the start and end bar/beat to be copied.
- Press ***Enter•Yes***. The display reads KEEP=OLD/NEW.
- Press ***Enter•Yes*** to keep the new track (or ***Cancel•No*** to keep the original track).

CMD TRACK	ERASE/UNDEFINE TRACK
	Press Command / Track / 2

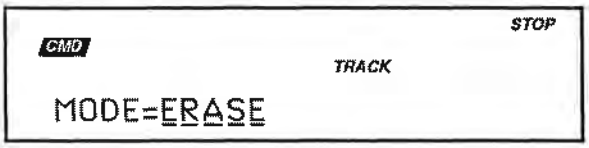
Use this command to erase a selected track. Keep in mind, even if you delete an instrument from an ***Instrument•Sequence Track*** location, the sequence/song track data is still there until you erase it. You can determine whether a track is defined by pressing ***Edit***, then ***Track***. Scroll to the Track Status parameter. An asterisk indicates that a track is undefined.

- Select ERASE/UNDEFINE TRACK.
- Press ***Enter•Yes***. The display reads TRACK=##. Select the track you want to erase by pressing one of the active ***Instrument•Sequence Track*** buttons.
- Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to abort the procedure).

CMD TRACK	**FILTER EVENT**
	Press Command / Track / 3

This command provides a convenient way to eliminate (ERASE) or copy specific information in a track, while leaving other track data intact.

- Select FILTER EVENT.
- Press ***Enter•Yes***. The display reads TRACK=##. Select the track you want to edit by pressing one of the active ***Instrument•Sequence Track*** buttons.
- Press ***Enter•Yes***. The display shows the following:



On this screen, by using the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons, you select the type of event to be "filtered" (ERASED or COPIED). Editing this parameter will change the Event type display, found by pressing the ***Right Arrow*** button. When MODE=COPY, the Event type display reads "COPY TO TRACK (##)." When MODE=ERASE, the Event type display reads "ERASE= (event type)."

The following event types are available for editing in this parameter:

- CNTL-1.MODWHEEL — Mod Wheel
- CNTL-4.PEDAL — Control Voltage
- CNTL-7.VOLUME — Volume
- CNTL-64.SUSTAIN — Sustain
- CNTL-70.PATCHSEL — Patch Select
- CNTL-XX. XCTRL — External Cnt
- ALL.CONTROLLERS
- CHANNEL-PRESSURE
- MIXDOWN-VOLUME — Track Volume
- MIXDOWN-PAN — Track Pan
- PITCH-BEND — Pitch Bend Wheel
- KEY-PRESSURE — Poly-Key Press
- PROGRAM-CHANGE
- KEY-EVENTS

Select the type of event for editing by using the ***Up/Down Arrow*** buttons, then press ***Enter•Yes***. At this point, there are two editing options:

- ERASE — This option will erase all events or a portion of events of the type specified. Press ***Enter•Yes*** and the display will read ENTIRE TRACK. Pressing ***Enter•Yes*** at this point will erase all of the selected events on the entire track and place you on the Audition page. If you wish to erase only events in a portion of the track only, press the ***Up Arrow*** button until the display reads SET RANGES. You can then press ***Enter•Yes*** and set both the key range and bar/beat range in which the editing will take place. After setting these ranges, the editing will take place and you will be placed on the Audition page. If you've previously set editing ranges for the current song or sequence, you can use the ***Up Arrow*** button to call up USE RANGES. Pressing ***Enter•Yes*** will commence editing based on the previously set range values.
- COPY — This option allows you to copy events to another track. Scroll right to select the Track numnber value. Use the ***Instrument•Sequence Track*** buttons to select the destination track. After selecting the destination track, press ***Enter•Yes***. From this point on, the editing proceeds from this point in the same manner as erasing events.

CMD TRACK	**MERGE TWO TRACKS**
	Press Command / Track / 4

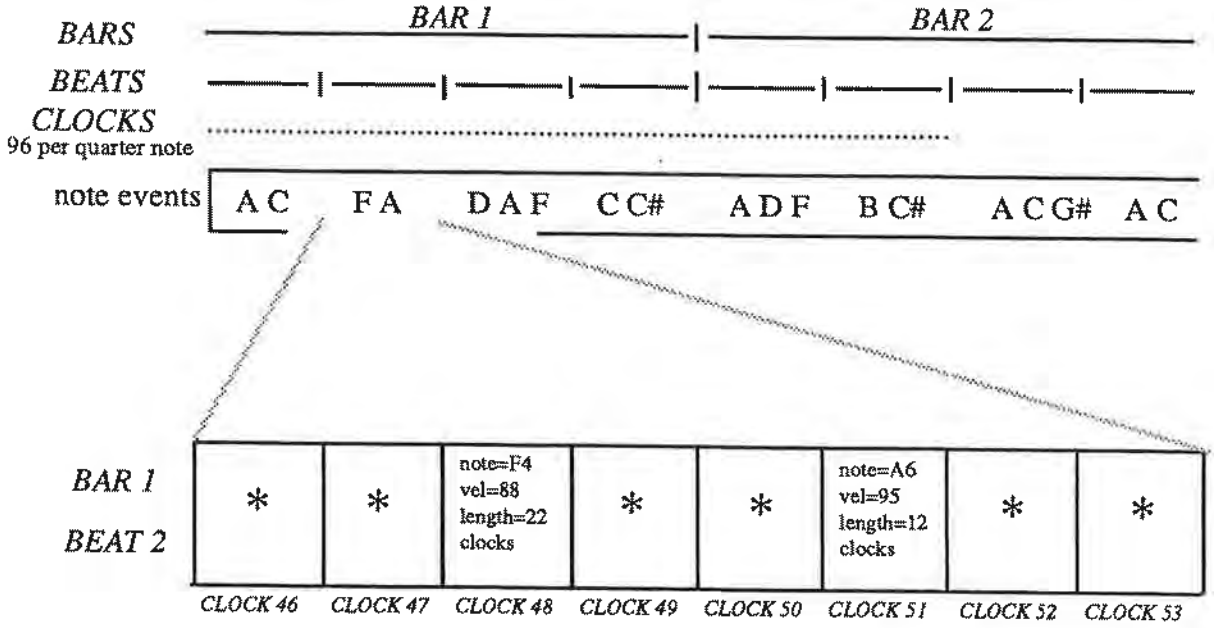
This command merges all key and controller information from one track into another track. This allows you to record several different tracks, all with the same instrument and MIDI configuration, then merge them into one. This is useful for creating sound-on-sound type sequences where you add rather than replace track data.

Note: You may run into problems if you try to merge two tracks with similar controller events. For instance, if you have pitch bend in your source track, and pitch bend in the destination track, and you merge them together at the *same point in the sequence*, it will result in "controller fights." If this happens, first erase the controller from one of the tracks, and then merge the two tracks together.

CMD TRACK	**EVENT EDIT TRACK**
	Press Command / Track / 5

Use this command to add, delete, lengthen, shorten, change the velocity, or the position of individual notes in a track. EVENT EDIT TRACK is a very useful command. Here are the fundamentals.

The sequencer uses "bars," "beats," and "clocks" to mark time during the recording of a track. The illustration below represents a stream of events recorded during two bars of a track.



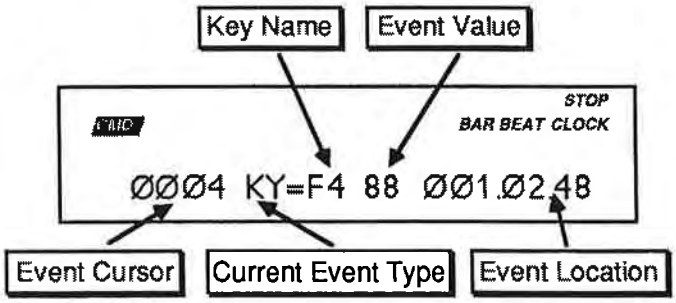
There are always 96 clocks per quarter note. The number of beats in a bar depends upon the time signature of the sequence. The number of bars depends upon the sequence length.

Sequencer tracks are made up of note events that occur at some location in the track. The location is defined in terms of bars, beats, and clocks. Note events consist of key, velocity, and duration information.

In the illustration above, an event occurs at BAR 1, BEAT 2, CLOCK 48. The note event is key F4. It was struck at a velocity of 88 (V=88). Its NOTE LENGTH (duration) is 22 CLOCKS. All this information is displayed for each key event when you use the EVENT EDIT TRACK command. You can change the key, velocity, or length of any key event. You also can delete events and insert new ones.

The EVENT EDIT TRACK function works on an individual track basis.

- With the track you want to edit selected, press the ***Command*** button, followed by the ***Track*** button. Scroll until the display shows EVENT EDIT TRACK. Press the ***Enter•Yes*** button. The display shows:



This is the Event Locating screen.

Event Cursor

When this field is selected, you can use the ***Data Entry Slider*** or ***Up/Down Arrow*** buttons to step ahead or back in time among the events in the track to find the one(s) you want to edit. When you press the ***Up/Down Arrow*** buttons, it will advance to the next or previous event of the type specified in the Event Type Filter. Moving the ***Data Entry Slider*** will scroll quickly through all the events in the track. You will hear the notes in the sequence play as you step through the track.

Current Event Type

This displays the current event type being viewed. This parameter is display only and cannot be changed from this screen.

Key Name

This parameter displays the key name that is either being played (when Current Event Type is "KY" KEY EVENTS) or being affected by pressure (when Current Event Type is "KP" KEY PRESSURE).

To change the Key Name for the current event, select the Key Name parameter and play the new key on the keyboard.

Event Value

This shows the value of the current event. The values can be changed using the ***Data Entry Slider*** or ***Up/Down Arrow*** buttons. These changes take place as they are made.

Event Location

Shows the current location in terms of Bar, Beat, and Clock. These three location fields can be individually selected by repeated presses of the ***Left/Right Arrow*** buttons. This allows you use the ***Up/Down Arrow*** buttons to step ahead or back in time by bars, beats or clocks, depending on which field is selected. Note that this parameter does not alter the track data in any way, it merely provides a method for locating and viewing events within a track.

Note: When event editing a song track, there will be an additional display which will have two more location values, corresponding to the step and repetition of the song. This screen can be found by pressing the ***Right Arrow*** button once.

Press the ***Left Arrow*** button until you see the following:

EVENT TYPE — Event Type Filter

This controls which types of events will be shown on the Event Locating screen as you scroll up through the events in the track. When set to AL = ALL EVNT, every event in the track will be listed. When set to any other value, only events of the type selected will be selected. Events of all other types will be left out of the list. When the Event Type is changed and you return to the Event Locating screen, the display will jump directly to the next event of the type shown, skipping other types. Event Types that can be selected here are:

MW MODWHEEL= Mod Wheel
FP FOOTPEDL= Foot Pedal
VL VOL-PEDL= Volume Pedal
SU SUSTAIN= Sustain Pedal
PS PATCH-SL= Patch Select
XC EXT-CONT= External Controller
CO ALL-CONT= All Controllers
CP CHANPRSR= Channel Pressure
MV MIX-VOL= Mixdown Volume
MP MIX-PAN= Mixdown Pan
PB PITCHBND= Pitch Bend Wheel
KP KEY-PRSR= Key Pressure
PC PROG-CHG= Program Change
KY KEY-EVNT= All Key Events
AL ALL-EVNT= All notes and controllers

Duration Screen for Key Events and Key Pressure Events

When event editing KY KEY-EVNTs or KP KEY-PRSR events on the Event Locating screen, pressing the ***Right Arrow*** button once (when editing a sequence track), or twice (when editing a song track) brings up an additional screen that looks like this:

These two parameters set the duration for a particular Key Event or Key Pressure Event in beats and clocks. Remember, this screen only appears when the current event is a KY Key Event or KP Key Pressure event.

INSERT and DELETE

After selecting a location with the Event Locating screen, you can INSERT or DELETE events.

- From the Event Locating Screen, press the ***Enter•Yes*** button. The display reads:

From this screen, you can use the ***Left/Right Arrow*** buttons to choose between the following:

1) INSERT a note, controller, or program change onto the selected track
2) DELETE a note controller, or program change from the selected track

- INSERT — This command is used to insert an event into the selected track. To utilize this function, select the type of event to be inserted with the Event Type Filter. Scroll to the Event Locating screen and select the location where the event is to be placed. On events other than key events, set the desired event value. Finally, use this screen to insert the event. Pressing ***Enter•Yes*** with INSERT underlined will insert the event.

- DELETE — Removes the event displayed on the Event Locating Screen. As with the INSERT command, select the type of event to be removed with the Event Type Filter. Scroll to the Event Locating screen and select the location where the event that is to be removed is located. Finally, use this screen to remove the event. Pressing ***Enter•Yes*** with DELETE underlined will remove the event.

Hint: When a Key Event is deleted, the key number and velocity are remembered. The next time a Key Event is inserted it will default to the value of the last Key Event deleted. This provides a convenient way to move a single Key Event that may be off-time. This only applies to KY KEY EVENTS.

Auditioning Changes

Once you have completed making changes to the track and would like to hear how they sound, press the ***Cancel•No*** button. This places you on the Audition page. From here, you can decide whether to keep the changes you've made, or go back and try again.

If you have not made any changes to the track, pressing ***Cancel•No*** exits the EVENT EDIT TRACK command.

CMD TRACK	**TRANSPOSE TRACK**
	Press Command / Track / 6

Use this command to transpose notes in a track by as much as an octave, up or down.

- Select TRANSPOSE TRACK.
- Press ***Enter•Yes***. The display reads TRACK=##.
- Select TRANSPOSE AMOUNT=## (Range -12 to +12 semitones).
- Select RANGE. Set the key and bar/beat ranges as described in earlier commands.
- Press ***Enter•Yes***. The display reads KEEP=OLD/NEW.
- Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to abort the procedure).

Hint: To transpose a track by more than one octave, use this command several times.

CMD TRACK	**SCALE EVENT**
	Press Command / Track / 7

Use this command to increase or decrease the effect of a controller in a track.

- Press ***Enter•Yes***. The display reads TRACK=##.
- Scroll to SCALE=__. The controllers on this page include:

KEY-VELOCITY	CNTL-XX.XCTRL
CNTL-1.MODWHEEL	ALL.CONTROLLERS
CNTL-4.PEDAL	CHANNEL-PRESSURE
CNTL-7.VOLUME	MIXDOWN-VOLUME
CNTL-64.SUSTAIN	MIXDOWN-PAN
CNTL-70.PATCHSEL	PITCH-BEND

- Scroll to SCALE BY ##. This value is a percent. The ***Data Entry Slider*** sets the whole number values; the ***Up/Down Arrow*** buttons set the fractional value. Values of less then 1.00 will decrease the effect of a controller. Values greater than 1.00 will increase the effect of a controller.
- Scroll to RANGE. Here you can select the entire range, or define a key range, and bar/beat to be scaled.
- Press ***Enter•Yes***. The display reads KEEP=OLD/NEW. Press the ***Play*** button to audition this event.
- Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to abort the procedure).

CMD TRACK	**SHIFT TRACK BY CLOCKS**
	Press Command / Track / 8

This command allows you to shift all note events on a track forward or backward in time. This can be useful for correcting MIDI delay, or creating a "lazy" or "rushed" feel when a track is shifted relative to other tracks.

- Scroll to SHIFT TRACKS BY CLOCKS.
- Press ***Enter•Yes***. The display reads TRACK=##.
- Scroll to SHIFT AMOUNT=## (Range -96 to +96 clocks).
- Press ***Enter•Yes***. The display reads KEEP=OLD/NEW.
- Press ***Enter•Yes*** to complete the command (or ***Cancel•No*** to abort the procedure).

Storage

Section 14 — Storage

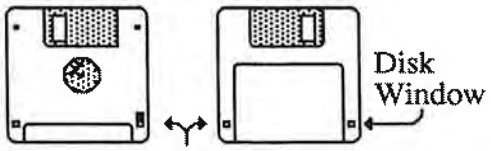
This section covers all data storage functions.

Disk Storage — Using the Disk Drive to Save and Load Data

The ASR–10's built-in disk drive is used to store all your Instruments, Banks, and Sequencer data, as well as System Exclusive messages from other MIDI devices. The ASR–10 uses a high-density (HD) disk drive, that can store 1600 Kilobytes of data on a Double-Sided High-Density (DSHD) 3.5" micro-floppy disk and 800 Kilobytes of data on a Double-Sided Double-Density (DSDD) 3.5" micro-floppy disk. The disks are enclosed in a protective plastic carrier with an automatic shutter to protect the diskette from physical damage. It is important not to alter this carrier in any way.

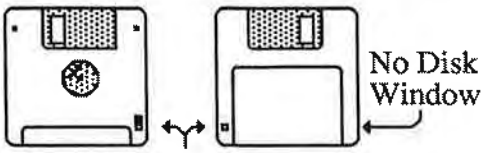
The 3.5" disks have a sliding write-protection tab so that you can protect your sounds and sequences against accidental erasure. Sliding the write-protection tab in the lower left corner of the disk so that the window is closed will allow you to store information on the disk. Sliding the tab so that the window is open will protect the disk against being accidentally reformatted or having files deleted. Double-Sided High Density disks can be easily identified because they have an additional window (with no write-protection tab) located on the lower right corner of the disk.

Double-Sided High-Density (DSHD)



Write Protect Tab

Double-Sided Double-Density (DSDD)



Write Protect Tab

Floppy disks are a magnetic storage medium, and should be treated with the same care you'd give important audio tapes. Just as you would use high quality audio tapes for your important recording needs, we recommend using high quality floppy disks for your ASR–10. You should keep your disks and the disk drive clean and free of dust, dirt, liquids, etc. Never put anything other than a disk in the disk drive. Do not expose disks to extremes of temperature or to strong magnetic fields. Also, never eject a disk while the drive is operating (the disk drive light is on).

ASR–10 Disk File Types

The ASR–10 disk storage system has been designed to allow maximum flexibility in saving, loading and organizing your instrument and sequence data. Consequently, there are several different types of files that can be saved:

- **INSTRUMENT** — individual ASR–10 instruments can be saved.
- **BANK** — contain "snapshots" of the currently loaded files.
- **SEQUENCE** — contains a single ASR–10 sequence.
- **SONG + ALL SEQS** — contains an entire song along with its related sequences.
- **EFFECT** — contains an effect algorithm that can be loaded to the BANK effect slot.
- **MIDI SYS-EX data** — In addition to the above types of ASR–10 data, you can use the disk drive to save system exclusive data from other MIDI devices. The ASR–10 can serve as the "librarian" for your other MIDI gear. This function is covered at the end of this section.
- **MACROS** — The file structure of the ASR–10 is set up so that each *directory* can hold up to 38 files. If any additional files are needed within the directory, a sub-directory must be created to hold those extra files. When working with a directory/file structure that is several layers deep, it is convenient to have a quick way to get to a sub-directory that is down deep in the structure. MACROS allow you to do that by using direct-dialing in much the same way that direct-dialing is used to access commands and parameters. The SAVE MACRO FILE command (on the Command/System•MIDI page) allows you to save your macros to disk.
- **O.S.** — The computer program that controls the ASR–10 (the *Operating System*, or O.S.) is contained on a disk and is loaded into RAM upon booting. The O.S. also contains the global parameter settings that are saved with the Command/System•MIDI SAVE GLOBAL PARAMETERS command. The disk that came with your ASR–10 contains the current O.S. Operating System upgrades will be available from your Authorized ENSONIQ Dealer as they are released. The COPY O.S. TO DISK command (on the Command/System•MIDI page) allows you to copy the O.S. to a different disk.

When you press the ***Load*** button and one of the ***Page*** buttons (***Instrument***, ***Seq•Song***, ***System•MIDI***, or ***Effects***), the ASR–10 displays any files of the particular type indicated that are available on the currently selected storage device. For example, on the Instrument Page, you will see a list of any Instruments or Bank Files which are present. The following chart shows which pages contain the different file types, and which indicator lights will be illuminated on the display for each type.

ASR–10 File Types:

MODE/Page	File Types	Indicators
Load/Instrument	Instruments	LOAD + INST
	Banks	LOAD + INST + BANK
Load/Seq•Song	Sequences	LOAD + SEQ
	Songs	LOAD + SEQ + SONG
Load/System•MIDI	Directories	LOAD + SYSTEM
	Macro Files	LOAD + SYSTEM + MACRO
	System Exclusive Data	LOAD + MIDI
Load/Effects	Effect Files	LOAD

Disk Capacity — Bytes, Blocks, and Files

The instruments, banks, songs, and sequences that the ASR–10 plays are stored on 3.5" micro-floppy disks. The ASR–10 has a high-density (HD) drive, allowing you to use both Double-Sided High-Density and Double-Sided Double-Density disks:

- **Double-Density disks** — can store 800 kilobytes of data, which translates into about 1600 *Blocks*.
- **High-Density disks** — can store 1600 kilobytes of data, which translates into about 3200 *Blocks*.

A Block is a handy unit which the ASR–10 uses to measure internal and disk memory — 1 Block=512 bytes=256 sample words; 2 Blocks=1k bytes; 4 Blocks=1k sample words.

ASR–10 disk files vary in size — how many will fit on a disk depends on the type of disk, the type of formatting, the number of files, and the size of the files.

There are two types of formatting that can be used to format a floppy disk:

- ENSONIQ — this format offers the largest amount of free blocks.
- COMPUTER — this format has fewer free blocks, because it allows Macintosh™ and IBM™ computers running the appropriate translation software to read the disks.

Disk Type	High -Density	High-Density	Double-Density	Double-Density
Format (Sector Offset)	ENSONIQ (offset 0)	COMPUTER (offset 1)	ENSONIQ (offset 0)	COMPUTER (offset 1)
Kilobytes	1600	1440	800	720
Sample Words	800k	720k	400k	360k
Blocks	3176	2863	1585	1426

There are limits to the number of files on a disk:

- Each disk can hold a *maximum* of 156 files.
- A disk can hold up to 60 files of any one file type, but no more than the total of 156 files of all types will fit on a single disk.

ASR–10 Disk Functions

FORMAT FLOPPY DISK — Formatting a Blank Disk

Before it can be used by the ASR–10 to store data, a disk must be formatted. Formatting puts information on the disk which the ASR–10 needs to read and write files. In addition to formatting a blank disk, the format procedure can be used to reformat a disk which has been used with some other device, such as a personal computer or another musical instrument. Note that any data (of whatever type) on a disk will be lost when the disk is formatted by the ASR–10.

To Format a floppy disk:

- Insert a blank, Double-Sided Double-Density or Double-Sided High-Density 3.5" micro-floppy disk into the disk drive, with the label-side facing up, and the metal shutter facing away from you. Make sure the plastic write-protect tab is in the closed position (no light showing through the window).
- Press ***Command***, followed by ***System•MIDI***.
- Press the ***Right Arrow*** button repeatedly until the following screen appears:

- Press ***Enter•Yes***. The display will read DISK LABEL=DISK000
  Use the ***Data Entry Slider*** and the ***Arrow*** buttons to give the disk a unique name by which you can identify that disk. This is very *important*, as the ASR–10 will ask you for this disk by name if an instrument or song on the disk is used in a bank. Also, make sure you write the Disk Label name on the outside of each disk after it has been formatted.
- Press ***Enter•Yes***. The display will read FORMAT TYPE=. Here you select which type of formatting you would like to use to format your floppy disk:

ENSONIQ — this standard format allows the ASR–10 to read the disk, and offers the largest amount of free blocks (DSDD disks format to 800k and DSHD disks format to 1600k).

COMPUTER — this unique format provides a lower disk capacity, but (having the first sector labelled sector 1) allows Macintosh and IBM computers that are running the appropriate translation software to read the disks.

- After choosing a format type, press ***Enter•Yes***.
- The display will read ERASE AND FORMAT DISK?
- Press the ***Enter•Yes*** button to proceed (pressing the ***Cancel•No*** button will return you to the Command/System•MIDI page).
- While the ASR–10 is formatting the disk, the display reads *FORMATTING* The formatting process takes between 80 and 160 seconds, depending on the format and disk type.
- When the formatting is done, the display reads DISK COMMAND COMPLETED, and then you are returned to the Command/System•MIDI page. The disk is ready to have ASR–10 files stored on it.

FORMAT FLOPPY DISK Messages:
- DISK DRIVE NOT READY — No disk in the drive.
- DISK FORMAT FAILURE -- DISK IS UNUSABLE. This indicates a defective disk. If you get this message, we advise that you throw out the disk in question. Try again with another disk.
- DISK WRITE-PROTECTED — The plastic write-protect tab in the lower-left corner of the disk must be closed (so you can't see through the hole) before anything can be written to the disk. Close the write-protect tab and try again.

COPY FLOPPY DISK — Making a Backup Copy of a Disk

The COPY FLOPPY DISK function lets you duplicate the contents of one entire disk (the *source disk*) onto another disk (the *destination disk*). This function will only work if both disks are the same size (**both** High-Density disks or **both** Double-Density disks). It is a good practice to regularly back up your valuable data — including the O.S. disk which came with the ASR–10 — in this way. You can use either blank or formatted disks. During the COPY FLOPPY DISK command, the ASR–10 will ask you if you want to format the blank disk. Make sure the formatted destination disks do not contain any files that you want to keep because they will be erased. It is also a good idea to *write-protect* the source disk, so there's no chance of losing data.

Note: This procedure uses the internal memory to temporarily hold the information while copying it between disks. The fewer files you have loaded in the internal memory before you start, the fewer times you will have to swap disks during the copy procedure. You will only have to swap disks if the size of the source disk you are copying is greater than the amount of free internal memory.

To Make a Backup Copy of a Disk:
- Slide open the plastic write-protect tab on the source disk (the original) so that the disk is write-protected (you can see through the hole). This is an extra precaution to safeguard the data.
- Press ***Command***, then ***System•MIDI***.
- Press the ***Right Arrow*** button repeatedly until the display shows COPY FLOPPY DISK.
- Press ***Enter•Yes***. The display shows INSERT SOURCE DISK (the one you want to copy).
- Insert the source disk, then press ***Enter•Yes***.
- The drive will engage and the display will flash READING SOURCE DISK. Once the drive stops, the display will change to show INSERT DEST DISK.
- Insert the destination disk (the one you want to copy to), and then press ***Enter•Yes***. If the destination disk is unformatted, the display will ask ERASE AND FORMAT DISK? Press ***Enter•Yes*** to format the disk. When formatting is complete, the drive will engage and the display will flash WRITING DEST DISK.

- After writing to the destination disk, the display will read VERIFYING DEST DISK. If the copy is complete, then the display will read DISK COMMAND COMPLETED.
- If the size of the source disk you are copying is greater than the amount of free internal memory, the display will change to show INSERT SOURCE DISK again. Repeat the preceding four steps as prompted by the ASR–10 display.

Note: When you copy a disk, the Disk Label Name is copied from the source disk to the destination disk.

COPY FLOPPY messages:

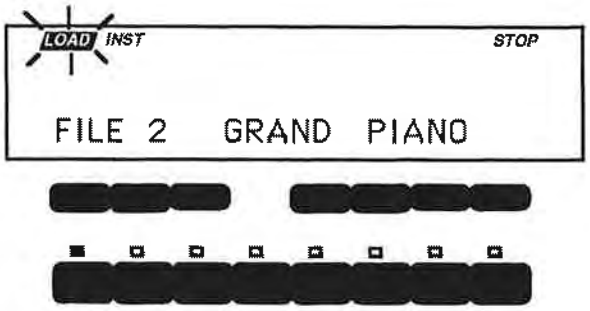
- DISK WRITE-PROTECTED — When asked for the destination disk you inserted a write-protected disk. The destination disk must have the write-protect tab closed (so you can't see through the hole).
- DISK IS NOT SOURCE DISK or DISK IS NOT DESTINATION DISK — You put the wrong disk in the drive when prompted for a certain disk. This is not fatal; it doesn't abort the copy procedure. Just insert the requested disk and proceed.

Loading and Saving ASR–10 Instruments

Loading an Instrument

You can load up to eight different instruments into the ASR–10 at once (within the limits of memory). First, insert a disk containing one or more instrument files into the disk drive.

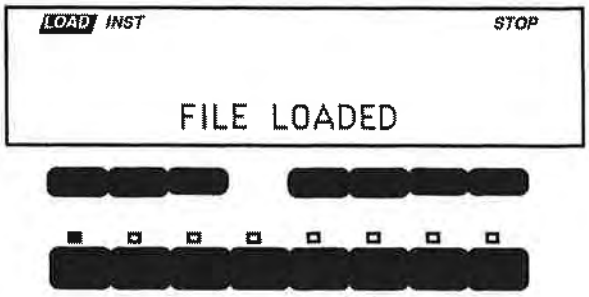
- Press ***Load***. The LOAD indicator flashes.
- Press ***Instrument***. The display might look like this:



When the LOAD indicator is flashing, the ASR–10 is showing you disk files (think of it as a question mark - the ASR–10 is saying "Load the file showing on the display?"). Pressing the ***Up/Down Arrow*** buttons takes you through the files on the disk. If there are none, the display will read "NO INST OR BANK FILES."

Whenever a disk file is displayed as above, you can press the ***Left*** or ***Right Arrow*** button to see the size of that file in *Blocks* (a Block is 256 samples; 4 Blocks=1K samples). Press the ***Left*** or ***Right Arrow*** button again to return to the file name.

- Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to view the various instrument files on the disk. Each file has its own File Number. When an instrument file is showing, the INST indicator is lit. The BANK indicator will light when a bank file is showing.
- Find the instrument you want to load, and press ***Enter•Yes***. The display will say PICK INSTRUMENT BUTTON. The ASR–10 is asking in which instrument location you want to load the sound into.
- Press any of the eight ***Instrument•Sequence Track*** buttons. The ASR–10 will begin immediately loading the instrument into the selected location. The display reads LOADING FILE... and the LED *flashes* red while the instrument is being loaded.



Once the instrument has been loaded, the display briefly shows "FILE LOADED."

The LED above the ***Instrument•Sequence Track*** button stops flashing and remains solidly lit red, indicating that there is now an instrument loaded in that location which can be selected by pressing that button.

If you tell the ASR–10 to load an instrument into a location that already has an instrument loaded (LED lit red), the new instrument will be loaded into that location and the one that was there will be automatically deleted.

If You Run Out of System Memory

You might have to delete an instrument(s) before loading the new one. If there are already one or more instruments loaded into the ASR–10, there might not be enough free memory to load the new one. In this case, the display will say PICK INST TO DELETE. At this point you can:

- Press any loaded ***Instrument•Sequence Track*** button. That instrument will be deleted from memory and the new one will be loaded; or,
- Press ***Enter•Yes***. The ASR–10 will delete an instrument (or instruments, as needed) for you, starting from the *highest-numbered* one in memory; or,
- Press ***Cancel•No***. The load command will be canceled with no harm done.

Note: You can continue to select and play existing instruments while the new one loads. No more "down-time" waiting for the next sound to load. With the ASR–10, you can be loading the next sound you need while continuing to play the currently selected sound.

Saving an Instrument to Disk

After you have created a new ASR–10 instrument, or made changes to an existing instrument, you can save the instrument to a formatted ASR–10 disk with the following steps:

- Insert a formatted disk into the drive. If you don't have a formatted disk, format one before proceeding.
- Press ***Command***, then ***Instrument***. This puts you on the Command/Instrument page, that contains all the instrument-related commands. Check to make sure that the instrument you want to save is selected. If not, press its ***Instrument•Sequence Track*** button to select it.
- Press the ***Left*** or ***Right Arrow*** button until the display reads SAVE INSTRUMENT (or use this shortcut — after pressing ***Command***, double-click on the ***Instrument*** button).
- Press ***Enter•Yes***.
- Edit the Instrument Name (if needed):

The display shows the current name with a cursor (underline) beneath the first character. If you want to give the instrument a new name, do so at this time. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the underlined character, then press the ***Left*** or ***Right Arrow*** button to move the underline. Repeat until the

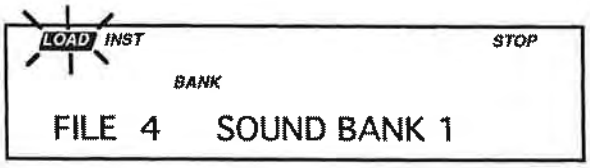
display shows the name you want (if you don't need to rename the instrument, just skip this step).

- Press ***Enter•Yes***.
  The display will read SAVING <INST NAME> while the instrument is being saved. Note that you can continue playing the keyboard while the instrument is being saved, and you can select other instruments. You cannot get out of Command mode until the save procedure is done.
- If there is already an instrument file with the same name on the disk, the display will ask DELETE OLD VERSION? Press ***Enter•Yes*** to save the instrument, replacing the one on the disk. This is for updating instruments to which you have made changes. Or press ***Cancel•No*** to abort the procedure.
- If there is not enough free space on the disk for the instrument, the display will say NOT ENOUGH DISK SPACE. Save the instrument to another disk (or delete some files from the disk).

Loading and Saving ASR–10 Banks

To Load a Bank

- Press ***Load***, then ***Instrument***.
- Use the ***Up/Down Arrow*** buttons to view the different files.



When a bank file is showing, the BANK indicator lights on the display along with the INST indicator.

- Once a bank file is showing, press ***Enter•Yes***. The ASR–10 will begin loading the instruments and the song data. As it loads each instrument the display tells you what it's doing. As soon as any of the instruments are finished loading, you can select that instrument and play while the rest of the bank loads.

At some point in the loading process, you may encounter the following message:

This prompt is informing you that an instrument or song saved as part of the bank is on a different disk than the disk currently in the drive. When this occurs:

- Eject the current disk in the disk drive and replace it with the disk that matches the Disk Label ID requested by the ASR–10 (you are writing these names on your disks, aren't you?).
- Press ***Enter•Yes***. The ASR–10 will resume loading until completed, or until the point when it needs another disk.
- When it has finished loading the instruments, the ASR–10 will load the song (if any) and then set up any copied instruments included in the bank.

Saving the Contents of Memory as a Bank

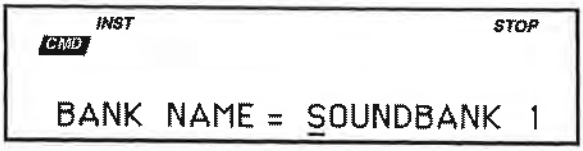
Saving a bank is like taking a "snapshot" of the contents of the ASR–10 memory. When you later load a bank, the ASR–10 "looks" at that snapshot and tries to recreate what was in memory when the bank was saved. You can use banks to automatically load a new group of instruments and/or a new song.

Banks are valuable for this reason: ASR–10 songs and sequences have no way of knowing which instruments are loaded into various ***Instrument•Sequence Track*** locations. Suppose, for example, you record a track with a piano instrument, but then load a kazoo sample into the instrument location where the piano was. The track will now play on the kazoo. By using banks you load instruments into their "proper" location. The Bank file stores the following information:

- Which instruments are loaded into each ***Instrument•Sequence Track*** location
- Which song and its related sequences are loaded into the internal memory
- Up to eight discrete Performance Presets
- The Bank Effect and its parameter settings
- The Edit/Track MIX, PAN, OUT, and EFFECT MOD CONTROL setting for each ***Instrument•Sequence Track***

It's easy to save the contents of the ASR–10 memory as a bank. Just remember that any instruments in the internal memory, as well as the song, must be saved separately before they can be saved as a bank.

- Save any instruments in memory to disk using the SAVE INSTRUMENT command on the Command/Instrument page.
- Save the song to disk (if you want a song as part of the bank) using the SAVE SONG + ALL SEQS command on the Command/Seq•Song page.
- Press ***Command***, then ***Instrument***.
- Press the ***Left*** or ***Right Arrow*** button until the display reads SAVE BANK.
- Press ***Enter•Yes***. The display shows:



The display shows the current name with a cursor (underline) beneath the first character. If you want to give the bank a new name, do so at this time. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the underlined character, then press the ***Left*** or ***Right Arrow*** button to move the underline. Repeat until the display shows the name you want. (If you are updating an existing bank, don't rename the bank; just press ***Enter•Yes*** and then press ***Enter•Yes*** once more in response to the question DELETE OLD VERSION?).

- Press ***Enter•Yes***.

Note: It's not necessary for the instruments and song to be saved on the same disk as the bank. Each instrument, the song, and bank can be saved on separate disks, but make sure the disks are labelled (WRITE DISK LABEL command found on Command/System•MIDI page) so that the bank can locate the information properly.

Loading and Saving Sequencer Data

There are two ways that ASR–10 Sequencer data can be stored on a disk:

- **SONG File.** A song file contains a song *and* all its related sequences. Loading a song file from disk will completely erase the current contents of the ASR–10 sequencer memory, replacing whatever is there with the song and sequences from the disk file.
- **SINGLE SEQUENCE File.** This type of file contains just one sequence. Loading a single sequence file will *not* erase the sequencer data already in memory — the new sequence simply becomes one more sequence in the internal memory which can be selected or used as a step in a song. There can be up to 80 sequences in memory at once.

Loading a Song or a Single Sequence from Disk

Both types of sequencer files are accessed from the Load/Seq•Song page.

- Press ***Load***, then press ***Seq•Song.***
  The LOAD indicator flashes, meaning that the ASR–10 is showing you disk files.
- Press the ***Up*** or ***Down Arrow*** button until the file you want to load is showing on the display:

When a SONG file is showing, the SONG indicator lights next to the SEQ indicator.

When a single SEQUENCE file is showing, only the SEQ indicator lights.

- Press ***Enter•Yes*** to load the sequencer file showing on the display. After you have loaded a song or sequence, it is automatically selected. Just press ***Play*** to play it.

Note: Remember that loading a song or sequence does not ensure that the right instruments are loaded into the proper ***Instrument•Sequence Track*** locations. Each track of each sequence will play whatever instrument is in that location at the time. The way to make sure everything is in the right place is to save the contents of memory (including the song) as a bank.

Saving a Single Sequence to Disk

Use the SAVE CURRENT SEQUENCE command to save a single sequence.

- On the Edit/Seq•Song page, select the sequence you want to save.
- Insert a formatted disk into the drive.
- Press ***Command***, then press ***Seq•Song.***
- Press the ***Left*** or ***Right Arrow*** button until the display reads SAVE CURRENT SEQUENCE.
- Press ***Enter•Yes.***
- Edit the Sequence Name (if needed):

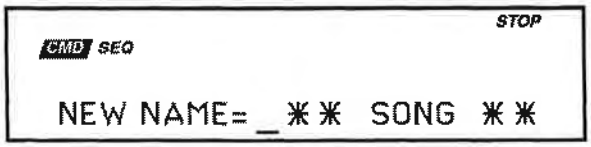
The display shows the current name of the sequence, with a cursor (underline) beneath the first character. If you want to give the sequence a new name, do so at this time. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the underlined character, then press the ***Left/Right Arrow*** buttons to move the underline. Repeat until the display shows the name you want (if you don't need to rename the sequence, just skip this step).

- Press ***Enter•Yes***. The display will read SAVING <FILE NAME> while the sequence is being saved.
- If there is already a sequence file with the same name on the disk, the display will ask DELETE OLD VERSION? Press ***Enter•Yes*** to save the sequence, replacing the one on the disk. Or press ***Cancel•No*** to abort the procedure.
- If there is not enough free space on the disk, the display will say NOT ENOUGH DISK SPACE. Save the sequence to another disk (or delete some files from the disk).
- If you try to SAVE CURRENT SEQUENCE while the song is selected, the ASR–10 will not execute the command, responding USE SAVE SONG + ALL. Again, make sure the sequence is selected before trying to save it.

Saving a Song (along with all Sequences) to Disk

Once you have created a song or made changes to an existing one, you can save the song to a formatted ASR–10 disk. In addition to saving the song itself, the SAVE SONG + ALL SEQS command saves all the individual sequences currently in memory (whether they are part of the song or not). To save a song:

- Insert a formatted disk into the drive.
- Press ***Command***, then press ***Seq•Song***.
- Press the ***Left*** or ***Right Arrow*** button until the display reads SAVE SONG + ALL SEQS.
- Press ***Enter•Yes***.
- Edit the song name (if needed):



The display shows the current name of the song, with a cursor (underline) beneath the first character. If you want to give the song a new name, do so at this time. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the underlined character, then press the ***Left/Right Arrow*** buttons to move the underline. Repeat until the display shows the name you want (if you don't need to rename the song, just skip this step).

- Press ***Enter•Yes***. The display will read SAVING <SONG NAME> while the song is being saved.
- If there is already a song file with the same name on the disk, the display will ask DELETE OLD VERSION? Press ***Enter•Yes*** to save the song, replacing the one on the disk. This is for updating songs to which you have made changes. Or press ***Cancel•No*** to abort the procedure.
- If there is not enough free space on the disk, the display will say NOT ENOUGH DISK SPACE. Save the song to another disk (or delete some files from the disk).
- Note that the song and all sequences in memory are saved as one file. You cannot extract a single sequence from a song file and load it separately. If you want access to a sequence individually, you must first load the song into the internal memory, and then save the individual sequence using the SAVE CURRENT SEQUENCE command.

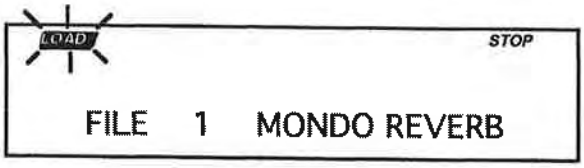
Loading and Saving ASR–10 Effects

Loading an Effect File

You can load an effect from disk into the internal memory of the ASR–10. The effect will load into the Bank effect location, replacing whatever effect was there previously.

First, insert a disk containing one or more effect files into the disk drive.

- Press ***Load***. The LOAD indicator flashes.
- Press ***Effects***. The display looks like this:



When the LOAD indicator is flashing, the ASR–10 is showing you disk files. Pressing the ***Up/Down Arrow*** buttons takes you through the effect files on disk. If there are none, the display will read NO EFFECT FILES.

Whenever a disk file is displayed as above, you can press the ***Left*** or ***Right Arrow*** button to see the size of that file in *Blocks* (a Block is 256 samples; 4 Blocks=1K samples). Press the ***Left*** or ***Right Arrow*** button again to return to the file name.

- Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to view the various effect files on the disk. Each file has its own File Number.
- Find the effect you want to load, and press ***Enter•Yes***. The ASR–10 will load that effect into the BANK effect slot. When finished, the display will read DISK COMMAND COMPLETED.

Saving an Effect to Disk

Use the SAVE BANK EFFECT command to save an effect to disk. If you wish to save either a ROM effect or an INST (Instrument) effect as a separate effects file, you must first copy the ROM or INST effect into the BANK effect using the COPY CURRENT EFFECT parameter on the Command/Effects page (see *Section 6 — EffectsParameters*) and then save it.

- Insert a formatted disk into the drive.
- Press ***Command***, then press ***Effect.***
- Press the ***Left*** or ***Right Arrow*** button until the display reads SAVE EFFECT.
- Press ***Enter•Yes.***
- Edit the effect name (if necessary):

The display shows the current name of the effect, with a cursor (underline) beneath the first character. If you want to give the effect a new name, do so at this time. Use the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the underlined character, then press the ***Left/Right Arrow*** buttons to move the underline. Repeat until the display shows the name you want (if you don't need to rename the effect, just skip this step).

- Press ***Enter•Yes***. The display will read SAVING <FILE NAME> while the effect is being saved.
- If there is already an effect file with the same name on the disk, the display will ask DELETE OLD VERSION? Press ***Enter•Yes*** to save the effect, replacing the one on the disk. Or press ***Cancel•No*** to abort the procedure.
- If there is not enough free space on the disk, the display will show NOT ENOUGH DISK SPACE. Save the effect to another disk (or delete some files from the disk).

A Note About Saving and Loading from a SCSI Storage Device

The preceding discussion has dealt with saving and loading ASR–10 files using the built-in 3.5 inch floppy disk drive. You can also use a SCSI storage device connected to the SCSI Interface on the rear panel to store instruments, banks, and sequence files. Much of the saving and loading procedure is similar to using the floppy disk drive.

Deleting a File from Disk

Deleting of disk files is done from Load mode. To delete (erase) a file from a disk:

- Press ***Load***, followed by the page button (***Instrument***, ***Seq•Song***, ***System•MIDI***, or ***Effects***) for the type of file you want to delete.
- Press the ***Up*** or ***Down Arrow*** button until the file you want to delete is showing on the display.
- While holding down the ***Load*** button, press ***Cancel•No***. The display will ask DELETE <FILE NAME>?
- Press ***Enter•Yes***. Press ***Cancel•No*** to abort this procedure.

MIDI System Exclusive Recorder

What is System Exclusive Data?

Some MIDI information—such as key events, controllers, program changes, etc. — is understood by virtually all MIDI devices, regardless of the manufacturer. The common ability to send and receive these messages is what allows you to play any MIDI device from any other, to change programs and volume remotely, to start and stop sequencers and drum machines together, and the many other performance miracles we have come to expect from MIDI.

There are other messages which each manufacturer has reserved for communicating specific information with a specific machine (or family of machines). These machine-specific messages are called *System Exclusive* (or Sys-Ex) messages, since they are meant to be recognized only by a particular device and ignored by all others (i.e. they are exclusive to a particular system).

System Exclusive Storage

It is not strictly necessary, however, for the receiving system to understand the data it receives, if the purpose is to store it for later reloading into the original system (just as it's not necessary for a file cabinet to understand the pieces of paper you file there). The ASR–10 can receive any MIDI System Exclusive message (up to 2 Megabytes standard/16 Megabytes with fully expanded memory) and save it to disk without having the slightest notion what it means or what type of device it came from. When you want to send the data back to the original device, you just load the data from disk back into the ASR–10, which will then retransmit the message exactly as it was received.

Note: System Exclusive messages cannot be stored across multiple floppy disks. System Exclusive dumps larger than 1600k can only be saved to a SCSI Storage Device.

Here are a few examples of the kinds of information which you can use the ASR–10 to store in this way:

- The program (patch) memory of virtually any MIDI synthesizer.
- The pattern memory a drum machine.
- The sequence memory of a MIDI sequencer.
- The preset memory of any MIDI reverb or other effects device which can send and receive it.

In short, any MIDI data (within memory limits) that can be transmitted from one device to another, can be received and stored by the ASR–10. With the ASR–10 at the heart of your system you now have disk storage for the data in all of your MIDI instruments.

Saving Sys-Ex Data Uses the Internal Memory

The ASR–10 uses the RAM (Random Access Memory) that is normally devoted to instruments, banks, and the sequencer to "buffer" incoming System Exclusive messages before saving them to disk. A buffer is an area of memory where data is held temporarily. When the ASR–10 receives a System Exclusive data dump, it stores it in the internal memory until you save the data to disk. As a result the contents of the ASR–10 internal memory is erased before the MIDI Sys-Ex recorder can be used.

Important:

You must save all instrument, bank, and sequencer memory before using the MIDI Sys-Ex recorder to receive data.

Loading Sys-Ex data from disk to an external device will not necessarily clear the internal memory. If there is enough unused sequencer memory to load and transmit the Sys-Ex message, the memory will not be affected. If there is not enough memory, you will be prompted to delete instruments until enough memory is available.

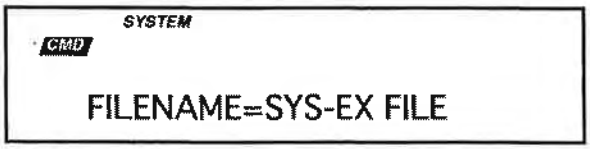
Saving System Exclusive Data from an External Device to Disk

Using the ASR–10's disk drive for storing data from external devices is a three-step process:

1) first get the ASR–10 ready to receive the data via MIDI;
2) next, send the data from the external device to the ASR–10; and finally
3) save the data to an ASR–10 floppy disk.

To save System Exclusive data from an external MIDI device:

- Connect the MIDI Out of the sending device to the MIDI In of the ASR–10.
- Insert a formatted 3.5" disk into the ASR–10 disk drive.
- Press ***Command***, then ***System•MIDI***.
- Press the ***Right Arrow*** button repeatedly until the display reads MIDI SYS-EX RECORDER.
- Press ***Enter•Yes***. The display shows the prompt MUST ERASE MEMORY OK? to warn you that any instruments, banks, sequences, and songs currently in memory will be lost. It's not too late, however, to quit and preserve the internal memory intact. Pressing ***Cancel•No*** at this point will return you to the Command/System•MIDI page without erasing the internal memory, allowing you to save it to disk before proceeding.
- Press ***Enter•Yes***. The display shows WAITING...
- From the external MIDI device, transmit the System Exclusive data. The display will flash RECEIVING... while the data is being sent.
- When the full message has been received, the display will read CANCEL=QUIT ENTER=SAVE to indicate that a complete message was recorded.
- Once you have successfully received the Sys-Ex message (or messages), press the ***Enter•Yes*** button to begin the process of saving the MIDI data to disk. The following display appears:



- Name the file with a 12-character name of your choice using the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the underlined character, and the ***Left/Right Arrow*** buttons to move the cursor. Repeat until the display shows the name you want.

- Press ***Enter•Yes***. The display reads SAVING <FILENAME> while the data is being saved. Or press ***Cancel•No*** to cancel the procedure for any reason.
- If there is a Sys-Ex file with the same name already on the disk, the ASR–10 will ask DELETE OLD VERSION? Press ***Enter•Yes*** to save the file, replacing the one on the disk. This is for updating files to which you have made changes. Or press ***Cancel•No*** to abort the procedure.
- After the file is saved, the ASR–10 returns to the FILENAME=<FILENAME> screen, so that you can save a backup copy of the data to a different disk if you wish. Press ***Cancel•No*** (or any other front panel button) to exit.

System Exclusive Messages

If during the reception of Sys-Ex data the ASR–10 display reads STATUS=INCOMPLETE, this means it did not receive an "End Of Exclusive" after the System Exclusive message. This would indicate that external device did not transmit a complete message for some reason. Press ***Cancel•No*** and try again.

If the ASR–10 display reads STATUS=MEMORY FULL, this means it ran out of memory with which to "buffer" the incoming data. After receiving the Memory Full message you can still save the file, but it is almost certain that the last Sys-Ex message received is incomplete and will not be able to be received successfully when retransmitted.

Loading System Exclusive Data from Disk to an External Device

After you have saved a System Exclusive message from an external MIDI instrument, getting the data back to the original instrument involves three steps:

1) First prepare the receiving instrument(s) to receive System Exclusive messages via MIDI;
2) next, load the data from the ASR–10 disk drive into memory; and then
3) send the data to the remote instrument(s) from the ASR–10.

To Load (and Transmit) a System Exclusive file from disk:

- Connect the MIDI Out of the ASR–10 to the MIDI In of the receiving device(s).
- Enable the receiving instrument(s) to receive System Exclusive messages. Many devices have a switch or a parameter which enables or disables reception of System Exclusive messages. Consult the manual for each particular device for details.
- Insert the disk containing the Sys-Ex data into the disk drive.
- Press ***Load***, then ***System•MIDI***.
- Use the ***Up/Down Arrow*** buttons to scroll through the files.
- Press ***Enter•Yes***. LOADING <FILE NAME> will appear on the screen as the ASR–10 loads the Sys-Ex data from its disk drive into internal memory. When completed, the display will look like this:

- Press ***Enter•Yes*** to send the data. The display will flash SENDING DATA... When completed, the display will return to the QUIT/SEND screen.
- Check the receiving instrument(s) to see that the data was received correctly. If it wasn't, make sure that:
  - ☛ the MIDI connections are correct (ASR–10 MIDI Out should be connected to the receiving device's MIDI In),
  - ☛ the receiving device is enabled to receive System Exclusive messages,
  - ☛ the receiving device is set to the same base MIDI channel as when the data was initially sent to the ASR–10.

  If all of this checks out, then press ***Enter•Yes*** again, and the data will be transmitted again. You can keep repeating this until the data has been successfully received by the remote device(s).
- Once the data has been successfully transmitted, press ***Cancel•No***.

ASR–10 Disk Messages

WARNINGS

Warnings are displayed for one second and indicate either successful completion or non-fatal error conditions encountered during a disk operation.

- DISK COMMAND COMPLETED — indicates that the disk operation was completed successfully.
- DISK WRITE-PROTECTED — appears during SAVE or DELETE operations if the diskette is write-protected.
- DISK HAS BEEN CHANGED — appears when you begin to LOAD, SAVE or DELETE when the disk in the drive has been changed since the last time the ASR–10 loaded a disk directory.
- DISK DRIVE NOT READY — usually indicates that there is no diskette in the drive, although it can indicate a hardware problem if it persists.
- NOT ENOUGH DISK SPACE — indicates that there are not enough available sectors on diskette or hard drive to save the file.
- FILE DOES NOT EXIST — appears when you try to LOAD or DELETE a file when the display reads FILE=NO <FILE TYPE>.
- NO SYS-EX DATA TO SAVE — indicates that there is no data in the Sys-Ex Recorder to save.
- FILE TOO LARGE TO LOAD — indicates that the sequence data in the disk file will not fit into the free memory. Delete sequences or songs to make some memory available. If that does not work, the file may have been saved from a system with more memory installed.

ERROR MESSAGES

Error messages are always accompanied by the PRESS ANY BUTTON TO CONTINUE... prompt, and remain on the screen until any button is pressed. These messages indicate serious error conditions which interrupted the disk operation. These errors may have prevented the correct saving or loading of the data in the file(s).

- DISK DRIVE NOT RESPONDING — indicates either that there is no diskette in the drive or that there are hardware problems.
- DISK NOT FORMATTED — the disk format was not recognized, and the disk is either blank or formatted for some other system.
- NOT ASR–10 DISK — the disk format was recognized, but the disk does not contain ASR–10 data.

- DISK ERROR - WRITE VERIFY — during a SAVE operation, the data written could not be verified. Probably indicates a bad disk sector. The file may be unusable.

- DISK ERROR - LOST DATA — during a disk read operation, the system missed data coming from the disk. Probably indicates a hardware problem. The file may be unusable.

- FILE OPERATION ERROR — one of a number of possible fatal errors occurred during a disk operation. Probably caused by a low-level hardware or software error, although the disk may be bad.

- DISK ERROR - BAD DATA — during a file operation, the CRC (error checking code) of the data block was incorrect. Probably indicates a bad disk sector. The file may be unusable.

- DISK ERROR - BAD DISK O.S. — during a file operation, the CRC of the Disk OS Program Control Block was incorrect. Probably indicates a bad disk sector. The disk may be unusable.

- DISK ERROR - BAD DIRECTORY— during a file operation, the CRC of the Directory blocks was incorrect. Probably indicates a bad disk sector. The disk may be unusable.

- DISK ERROR - BAD FAT — during a file operation, the CRC of the FAT (File Allocation Table) block was incorrect. Probably indicates a bad disk sector. The disk may be unusable.

- DISK ERROR - BAD DEVICE ID — during a file operation, the CRC of the Device ID block was incorrect. Probably indicates a bad disk sector. The disk may be unusable.

- FORMAT FAILED - BAD DISK — during formatting, a bad disk sector was detected. The disk may be defective, and should not be used.

- O.S. NOT ON DISK — the system was unable to LOAD the O.S. file. The file may be unusable.

- DISK COPY NOT COMPLETED — appears if ***Cancel•No*** is pressed during the Disk Copy procedure.

Sampling Applications

Section 15 — Sampling Applications

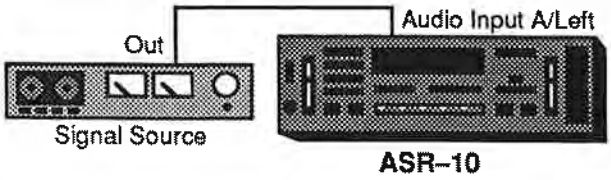
Resampling through the EQ Effect — Using Equalization

Often a sound contains some overtones or harmonics which sound normal on one note but unnatural when transposed over a range on the keyboard. Or it might be necessary, when trying to match up the different samples in a multisampled sound, to apply some drastic EQ to one or more of those samples. This is why the single most valuable tool for shaping a sampled sound probably is a good equalizer.

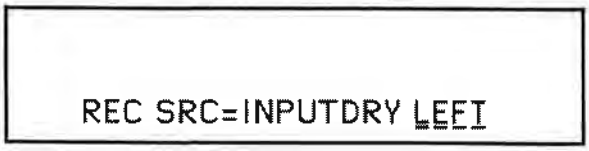
The ASR–10 offers an excellent parameteric EQ within its ROM effects that allow you to equalize the signal and resample the processed signal back into the ASR–10. Here's a little trick for finding the proper EQ settings:

Record a Dry Sample

The first step is to sample the signal source flat, without any EQ or other processing:



- To do this, press the ***FX Select•FX Bypass*** button and set it to FX=OFF 23 VOICE 44K for the best frequency response (remember that this is setting your sampling rate for sampling).
- Connect the sound source to the Left (and Right for stereo) Audio Input jack on the rear panel of the ASR–10.
- Press the ***Sample•Source Select*** button. The display shows:



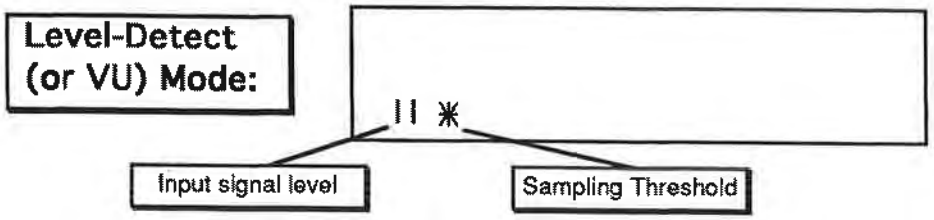
REC SRC= INPUTDRY LEFT is the default, allowing you to sample a dry signal connected to the Left Audio Input (in mono).

Note: If you want to sample in stereo (for instance, when sampling a stereo synthesizer), press the left ***Audio Track*** button, and while holding it down, press the right ***Audio Track*** button (or press the ***Up Arrow*** button twice) to select L+R. The display will reflect the Audio Track status.

For additional information about Selecting, Enabling, and Disabling Audio Tracks, see *Section 12 — Sequencer/Audio Track Concepts.*

Because we want to record a dry unprocessed signal, leave the Record Source set to INPUTDRY.

- Press the left ***Audio Track*** button to hear the input source (red LED lit).
- Press an unused (LED not lit) ***Instrument•Sequence Track*** button. The ASR–10 goes into Level-Detect (or VU) mode:



Once you have adjusted the input signal level:

- Press ***Enter•Yes*** or press the left foot switch to initiate sampling. The display will show "WAITING..." until the threshold is exceeded. The AMP indicator will light if the Input Source clips. During sampling, the display shows:

The xxx value (range 000 to 999) will count down in real time when sampling. Press any button to stop sampling. Remember that if the sampling threshold (the star) is set higher than the input signal level (the vertical bars), the ASR–10 will not begin sampling until the input signal crosses the threshold. Pressing ***Cancel•No*** while the sampling screen reads WAITING will abort sampling, and the ASR–10 will return to Level-Detect mode.

- Play the sound to be sampled; play the guitar, synth, start the recording, or speak into the microphone. If you are planning on using a microphone, we recommend plugging it into a mixer or preamp, and sending a line from the mixer/preamp to the Left Audio Input jack, for optimal performance.
- Press ***Cancel•No*** or press the Left Foot Switch to stop sampling. If you don't stop it by pressing ***Cancel•No*** or the Left Foot Switch, the ASR–10 will continue sampling until it has used up all the available memory.

Note: In order to use the foot switch to start and stop sampling, the LEFT FOOTSW parameter on the Edit/System•MIDI page must be set to LEFT FOOTSW= SAMPL YES.

- After you have stopped sampling (or the memory is full) the display flashes PLAY ROOT KEY OR ENTER. The Root Key is the incoming MIDI note number that will play the WaveSample at its original pitch.. In other words, if you play a C4 on the MIDI keyboard connected to the ASR–10, the sample will play back at "unity" (the same pitch as the original input signal) whenever the C4 key is pressed. Whichever key you press in response to the PLAY ROOT KEY OR ENTER prompt is the key that will play back exactly what you sampled.

Hint: You can also press the ***Enter•Yes*** button in response to the PLAY ROOT KEY OR ENTER prompt. This will set the root key of the new WaveSample to the same note assigned to the ENTER PLAYS KEY parameter, and allows you to record samples using your ASR–10 without being connected to a MIDI keyboard. Remember, you can assign different WaveSample ranges and change the pitch using the Edit/Instrument parameters.

Normalizing Gain

After you've recorded a satisfactory sample, you should then perform the Command/Amp NORMALIZE GAIN command, to optimize use of the full dynamic range of the ASR–10. To do this:

- Press ***Command***, then ***Amp*** and scroll until the display reads NORMALIZE GAIN.
- Press ***Enter•Yes*** to invoke the command. The display will read DATA BEING PROCESSED, followed by COMMAND COMPLETED.

Setting up the Parametric EQ Effect

Play the sample on various notes, listening to the overall tonal balance — does it need some added highs, or perhaps a little less midrange, etc. Listen particularly for frequencies that seem to stand out unnaturally when the sound is transposed.

- Press the ***FX Select•FX Bypass*** button, and use the ***Data Entry Slider*** or the ***Arrow*** buttons to select FX= ROM-49 PARAM EQ. The display shows:

- Press ***Edit*** followed by the ***Effects*** page button. The display shows the VAR= xxx screen. Play the ASR–10 at the root key of the new sample (it's very important that you play the root key during this process so that you'll be equalizing exactly the pitch you sampled). While playing the root key, adjust the PARAM EQ effect parameters (using the ***Left/Right Arrow*** buttons to select the parameters, and the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons to change the values), zeroing in on the problem areas and cutting or boosting those frequencies, until the sample sounds just the way you want (for a complete description of the Parametric EQ parameters, see *Section 6 — Effect Parameters*).

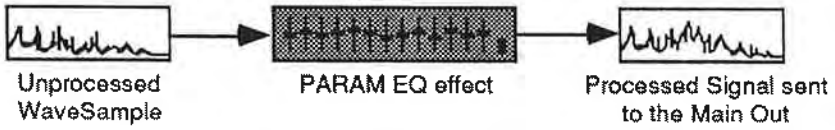
Resampling with Effect

- Without changing any of the settings on the PARAM EQ effect, press the ***Sample•Source Select*** button. The display reads REC SRC= INPUTDRY LEFT.
- LEFT (underlined) is the default, allowing you to resample the LEFT channel of the Main Output in mono.

Note: If you want to resample in stereo (for instance, when resampling with an effect that has stereo movement, like EQ+TREM+DDL), press the left ***Audio Track*** button, and while holding it down, press the right ***Audio Track*** button (or press the ***Up Arrow*** button twice) to select L+R. The display will reflect the Audio Track status.

- Press the ***Right Arrow*** button once to select INPUTDRY (underlined).
- Press the ***Up Arrow*** button twice to select MAIN-OUT. For the optimal signal-to-noise ratio, the Volume Slider should be at maximum (all the way up) when sampling the MAIN-OUT. The display looks like this:

With this set-up, you are sending the processed main output signal *back into* the ASR–10. In this routing, the processed WaveSample is theoretically sent out the Main Output of the ASR–10 and back into the Audio Input to be resampled (much like a studio effects loop).



- Press an unused ***Instrument•Sequence Track*** button. This puts you back into Level-Detect (or VU) mode.
- Press ***Enter•Yes*** and play the processed WaveSample to initiate sampling (as described earlier).

The EQ curve you set up in the previous step is now "built into" the sample, and it should sound just as it did when you were playing it through the PARAM EQ. To verify this, press the ***FX Select•FX Bypass*** button and scroll down to FX= OFF 31 VOICE AT 30K. You should hear the WaveSample with the EQ curve built into it.

If you find that you have boosted or cut any given frequency too much, just change the EQ settings slightly and sample the sound again. With a little practice you will be able to identify quickly problem frequencies and EQ them out using this method.

Hint: After sampling the MAIN OUT, the Edit/Amp BOOST parameter is automatically set to ON in order to match the level between the original WaveSample and the resampled WaveSample. As a final step, after you're happy with the resampled results, you can futher optimize the WaveSample by setting the Edit/Amp BOOST parameter to OFF, and performing the Command/Amp NORMALIZE GAIN command (as explained above).

Using the ASR–10 as a Virtual Multi-Track Digital Recorder

Because the ASR–10 has several unique sampling features (such as the ability to sample through the Effects and the MAIN OUT sample mode, which allows you to resample sequence playback or notes played on the MIDI keyboard), you can use the ASR–10 as a virtual multi-track digital recorder. Doing this requires some creative thinking on how to use some of these sampling features. This application note will give you some ideas and point you in the right direction.

- First, load instruments into ***Instrument•Sequence Track*** locations 1-6 and record a 4 bar sequence. Set the Effects the way you want them to be heard.
- The next step is to sample this sequence. To do this, set the ASR–10 to REC SRC= MAIN OUT L+R on the Sample•Source Select page. Press ***Enter•Yes*** to enter the sampling page and select one of the two unused ***Instrument•Sequence Track*** locations when you see the PICK SAMPLE INSTRUMENT prompt. Now initiate sampling. As soon as sampling has begun, start the sequencer. Let the sequence play through one repetition, then hit ***Cancel•No*** to end sampling. Select a root key when prompted, and you're ready for the next step.
- At this point, you may need to edit the SAMPLE START point of the sample. Here's a tip on doing this. Go back to your sequence, and press ***Record*** and ***Play***. On the first beat of measure 1, play the root key of the Instrument containing the new sample, and hold it down for the entirety of the four bar sequence. The sample will most likely not begin playing at the point where the music from your original sequence starts playing, but that's okay. QUANTIZE this track to 1/4 notes when you're finished.

The purpose of doing this is to give you a shortcut for finding the correct SAMPLE START location. Here's how:

- Select the new sample for editing and go to the SAMPLE START screen. Now, adjust the percentage parameter, but instead of striking the key each time to hear if you've found the right spot, hit the ***Play*** button of the sequencer. Press the ***Stop•Continue*** button if you don't hear music begin playing as soon as the sequencer starts playing, then reset the SAMPLE START parameter, and try again.

  Now, here comes the benefit of doing it this way. When you think the SAMPLE START point is close, start the sequencer again, this time with both the CLICK and LOOP parameters set to ON. You can now edit using the absolute sample start parameter. If you have found exactly the right SAMPLE START point, the click will be in time with the music. Also, when the sequencer loops, it will restrike the sample. You should hear no "glitch" when the sequence loops. Once you have your sample meeting both of these standards, you've found the right SAMPLE START location.

- Next, move on to the SAMPLE END parameter and set it to a point just past the end of the fourth measure of music. For this edit, you will have to strike the key, since playing the sequencer will never play the sample all the way to the end (it loops, remember?). Once you have a SAMPLE END point set, use the TRUNCATE WAVESAMPLE command to get rid of the unneeded data at the beginning and end of the sample.

- After playing your new sample, you can use the DELETE INSTRUMENT command to eliminate instruments 1-6. Remember, since our sequence is now a sample, you're able to erase all of the instruments used to make the sequence and in so doing, free up some memory.
- You can now record a second digital audio track. Plug in a guitar, bass, or any line signal into the Left Audio Input of the ASR–10. If you are planning on using a microphone, we recommend plugging it into a mixer or preamp, and sending a line from the mixer/preamp to the Left Audio Input jack, for optimal performance. Use the Input Level knob to set the correct level.
- Set the REC SRC page to INPUT+FX LEFT. Press ***Enter•Yes*** to enter sampling mode and select an unused ***Instrument•Sequence Track*** when you see the PICK SAMPLE INSTRUMENT prompt.
- Next, play the sequence and play or sing along with it until you feel comfortable with your new part. Then, press ***Enter•Yes*** again to begin sampling. As soon as sampling has begun, start the sequencer and play or sing along through one 4 bar repetition. Then, hit ***Cancel•No*** to stop sampling. Select a root key when prompted.

  Notice that when you play back the new sample, you hear only the new part. This is because the sequencer is not recorded in REC SRC= INPUT+FX (or REC SRC= INPUTDRY) mode. The sequence was only acting as a timing guide for the new part. This allows you to have track separation to set Volume and Panning information.
- Now, record this new sample into the sequence by playing on the first beat of bar 1, then QUANTIZE to 1/4 notes, just like you did with the first sample. You can then proceed to edit the SAMPLE START and end parameters, as described above. When the SAMPLE START time is correct, you'll hear this new track "sync up" with the original sample.
- You can now continue adding tracks in this fashion until you get close to running out of memory, or have only one sequence track left. Once either of these things happens, go on to the next step.
- Now you can "bounce down" the tracks you have to a stereo mix. To do this, set your track mix and pan levels where you'd like them. You can also add a new Effect onto the entire mix, if you desire. Then, set the REC SRC parameter to MAIN OUT L+R. Press ***Enter•Yes*** and select an unused ***Instrument•Sequence Track*** location when you see the PICK SAMPLE INSTRUMENT prompt. You are now essentially back at the beginning of this application note, taking a sequence of multiple tracks and sampling it into a two track master mix. If you'd still like to add more tracks, you can again erase all of the instruments that were in the sequence you just sampled, and proceed repeating the above steps in the application note.

Happy Recording!

Instrument Programming Applications

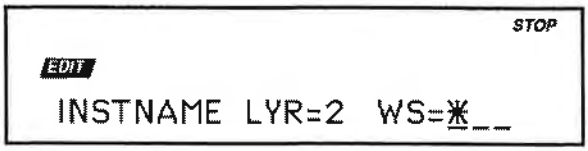
Section 16 — Instrument Programming Applications

This section provides information on some of the most commonly used Instrument programming applications.

Making a Stereo Sound from a Mono WaveSample

Within an Instrument, you can take a mono WaveSample, copy it to another layer, and pan each WaveSample left and right to create a stereo sound. You can also select a layer (containing many WaveSamples), copy the layer, and then pan each layer left and right to create a stereo sound. For the purpose of this discussion, we'll copy a WaveSample from an Instrument that only has one layer, and one WaveSample. Here's how:

- Make sure the Instrument with the WaveSample to be copied is selected, by pressing it's ***Instrument•Sequence Track*** button (LED solidy lit yellow).
- Press ***Command***, followed by ***Layer*** and scroll until the display shows CREATE NEW LAYER.
- Press ***Enter•Yes***. The display momentarily reads LAYER 2 CREATED.
- Press the ***Edit*** button. The display shows:



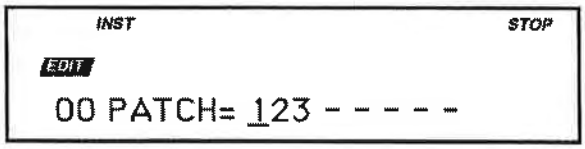
Here you see the currently selected instrument, the newly created layer (LYR= 2), and WS= * (the * indicates that there are no WaveSamples in this layer yet).

- Move the cursor (underline) to LYR = 2. The cursor should be beneath the Layer number. If it's not, use the ***Left Arrow*** button to move it there.
- Press the ***Up/Down Arrow*** buttons to select the first layer (LYR= 1).
- Move the cursor (underline) to WS = __. Use the ***Right Arrow*** button to move it there.
- Play the controller keyboard or press the ***Up/Down Arrow*** buttons to select the WaveSample to be copied.
- Press ***Command***, followed by ***Wave*** and scroll until the display shows COPY WAVESAMPLE.
- Press ***Enter•Yes***. The display shows TO INST= (Instrument name). Select the *same* Instrument.
- Press ***Enter•Yes***. The display shows TO LAYER=___. Select Layer 2.
- Press ***Enter•Yes***. The display shows either COPY= PARAMS ONLY or COPY=PARAMS +DATA. Since the WaveSample copy is going in the same Instrument, we don't need to copy the data, and choosing COPY= PARAMS ONLY uses the least amount of memory. Select COPY= PARAMS ONLY.
- Press ***Enter•Yes***. The display briefly shows WAVESAMPLE 2 CREATED.

Now that we have the same WaveSample copied to 2 different layers, we can pan each WaveSample left and right to create a stereo sample.

- Press the ***Edit*** button, and select (same Instrument name) LYR= 1 WS= 1 using the ***Arrow*** buttons as explained above.
- Press the ***Amp*** button and scroll until the display shows WS VOLUME=99 PAN=+0.
- Move the cursor (underline) to PAN= +0. The cursor should be beneath the pan value. If it's not, use the ***Left /Right Arrow*** buttons to move it there.
- Use the ***Data Entry Slider*** to set the value to PAN= +99.
- Press the ***Edit*** button, and select (same Instrument name) LYR= 2 WS= 2.
- Press the ***Amp*** button. The display shows WS VOLUME=99 PAN=+0.
- The cursor will still be beneath the pan value.
- Use the ***Data Entry Slider*** to set the value to PAN= -99.

You've just taken a mono WaveSample, copied it to another layer, and panned them left and right to create a stereo sound. If you aren't hearing both stereo signals, double-click the ***Instrument*** page button (the 00 PATCH= screen is displayed), and make sure both Layers 1 and 2 are selected.



If not, move the cursor with the ***Left/Right Arrow*** buttons and select/deselect the layers with the ***Up/Down Arrow*** buttons. In the above example layers 1, 2, and 3 are selected. Layers 4 through 8 are deselected. Take the time to enjoy your newly created stereo sound.

Hint: One of the most common tricks to creating a "chorused" effect is to slightly detune each channel of a stereo WaveSample in equal amounts of positive and negative values. Here's how to do this with our newly created stereo example:

- Press the ***Edit*** button, and select (same Instrument name) LYR= 1 WS= 1 using the ***Arrow*** buttons as explained above.
- Press the ***Pitch*** button and scroll until the display shows ROOT KEY=(key location) FINE=+0.
- Move the cursor (underline) to FINE= +0. The cursor should be beneath the fine tune value. If it's not, use the ***Left /Right Arrow*** buttons to move it there.
- Use the ***Up/Down Arrow*** buttons to set the value to FINE= +5.
- Press the ***Edit*** button, and select (same Instrument name) LYR= 2 WS= 2
- Press the ***Pitch*** button. The display still shows ROOT KEY=(key location) FINE=+0.
- Use the ***Left /Right Arrow*** buttons to set the value to FINE= -5.

For more of a "chorused" effect, set the positive/negative values to a wider range. For less of a "chorused" effect, set the positive/negative values to a closer range.

Creating Keyboard Cross Fades

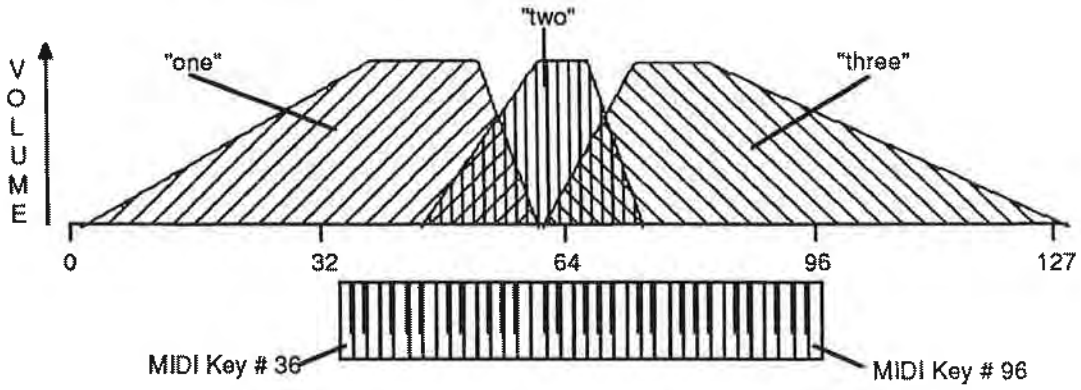
We'll use the Edit/Amp, Volume Modulator Cross Fade Breakpoints to create a keyboard cross fade. First, create three samples in three layers:

- With a microphone connected to the left Audio Input of the ASR–10, press ***Sample•Source Select***, set REC SRC= INPUTDRY LEFT, and then pick a sample instrument. Press ***Enter•Yes*** and say "one," then press ***Cancel•No*** to stop sampling. Make the lowest C on the keyboard the root key for sample "one" in layer 1.
- The next sample will be in a NEW layer. Press ***Sample•Source Select***, then pick the same sample instrument. Underline LYR= 1 and change that to LYR= N. Press ***Enter•Yes*** twice. Say "two." Press ***Cancel•No*** and assign the root key to middle C (two octaves above the lowest C). That's sample "two" in layer 2.
- Press ***Sample•Source Select***, then pick the same sample instrument. Underline LYR=2 and change that to LYR=N. Press ***Enter•Yes*** twice, and say, "three" into the microphone. Press ***Cancel•No*** and assign the root key to the C two octaves above middle C. Sample "one" is in the bottom third of the keyboard. Sample "two" is in the middle third, and sample "three" is in the top third.

Press the ***Edit*** button to get to the Edit Context page. Adjust the parameters in the display so they read UNNAMED LYR=1 WS=ALL. Press the ***Amp*** button. You are now on the Edit/Amp page. Scroll right to VOLUME MOD and adjust the parameters so the display reads VOLUME MOD=KBD * 0. This selects KBD (keyboard) as the volume modulator. Underline the number in the display and change it to 99. The display will read VOLUME MOD=KBD * 99. Scroll right to A-B FADE IN and set the parameters in the display to read, A-B FADE IN=0 TO 40. Scroll right one more page, and set the parameters in the display to read, C-D FADEOUT=50 TO 60. Press the ***Edit*** button to return to the Edit Context page. Underline LYR=1 in the display and change that to LYR=2. The display now reads UNNAMED LYR=2 WS=ALL.

Press the ***Edit*** button to return to the Edit/Amp page. Scroll right to the VOLUME MOD page, and set it to VOLUME MOD=KBD * 99. Scroll right to the A-B FADE IN page and set it to A-B FADE IN =45 TO 55. Scroll right to the C-D FADE OUT page and set it to C-D FADE OUT=65 TO 75.

Press the ***Edit*** button and set the parameters in the display to read UNNAMED LYR=3 WS=ALL. Press the ***Edit*** button again to return to the Edit/Amp page. Scroll right to VOLUME MOD, and set it to VOLUME MOD=KBD * 99. Scroll right to A-B FADE IN and set the parameters to A-B FADE IN=60 TO 70. Scroll to C-D FADE OUT and set it to C-D FADE OUT=80 TO 127. The following diagram illustrates the keyboard crossfade that you have now constructed:



As you play chromatically up the MIDI keyboard connected to the ASR–10 you will hear the three different WaveSamples fading in and out. The voice samples could have been piano or synthesizer samples, and the result would have been the same. When the keyboard is used as the Volume Modulator, the Crossfade Breakpoint parameters enabled you to blend the samples together seamlessly.

Note that when the keyboard is used as the volume modulator, the numbers selected for A, B, C, and D refer to MIDI key numbers. The low C on the ASR–10 is MIDI key # 36; the high C is MIDI key number 96. Middle C is MIDI key # 60. And so on. (The keys of an 88-note piano-style keyboard go from MIDI key 21 to MIDI key 108.)

Although the crossfade described above uses the keyboard to determine the breakpoints. The output of any of the available modulators found on the Edit/Amp Volume Mod parameter can be used.

See *Section 11 — WaveSample and Layer Parameters* for more information on the Volume Modulator Cross Fade Breakpoints.

Velocity Cross-Switching Between Layers

This application will give you an example of how velocity can be used to call up layers:

- Press the ***Sample•Source Select*** button, set REC SRC= INPUT DRY LEFT, and then pick a sample instrument. Press ***Enter•Yes*** and say "one," then press ***Cancel•No*** to stop sampling. Make middle C the root key. This is layer one.
- Press the ***Sample•Source Select*** button again, pick the same sample instrument. The display reads UNNAMED LYR=1 WS=NEW. Underline LYR=1, and set it to LYR=N. Press ***Enter•Yes***. Press ***Enter•Yes*** again and say "two." Again, make middle C the root key. This is layer two.
- Press the ***Sample•Source Select*** button, then pick the same sample instrument. Underline LYR=2 in the display, and change it to LYR=N. Press ***Enter•Yes*** twice, and say "three." This is layer three. Again, select middle C as the root key. When you play middle C, you will hear samples, "one, two, and three," in layers one, two, and three.

Now that we have our WaveSamples, the Edit/Layer LAYER VELOCITY parameter can be used.

- Press ***Edit***. This places you on the Edit Context page. Using the ***Data Entry Slider*** or ***Arrow*** buttons, adjust the parameters on the page so the screen reads, UNNAMED LYR=1 WS=ALL.
- Press the ***Layer*** button. This will take you to the Edit/Layer page. Scroll to the LYR VEL screen and underline and adjust this parameter so it reads LYR VEL LO=0 HI=42.
- Press the ***Edit*** button again, which takes you back to the Edit page. Underline LYR=1 in the display and change it to LYR=2. The display reads UNNAMED LYR=2 WS=ALL. Pressing the ***Edit*** button takes you back to the Edit/Layer page (the ***Edit*** button acts as a toggle).
- Set the Edit/Layer parameters to LYR VEL LO=42 HI=85. Press the ***Edit*** button again, to return to the Edit Context page. Underline LYR=2 and set it to LYR=3. Press the ***Edit*** button to toggle back to the Edit/Layer page. Set these Parameters to LYR VEL LO=86 HI=127.

All three Layers are now set to respond to velocity. A layer will only play when the note-on velocity is within the range specified on this page. A soft attack will call up sample "one," a medium attack, sample "two," and a hard attack, sample "three." Remember, you can do this with up to eight layers.

Using the ENTER PLAYS KEY Parameter

The ENTER PLAYS KEY parameter, found on the Edit/System•MIDI page, allows you to play any key between A0 to C8 without having a MIDI Controller connected to your ASR–10. By using the ***Data Entry Slider*** or the ***Up/Down Arrow*** buttons, you can select each key and play it by pressing the ***Enter•Yes*** button. This is useful for hearing the different WaveSamples in a multisampled instrument (like a drum kit for instance).

Let's experiment with this feature by using the ***Enter•Yes*** button to play some of the keys assigned in the STEREO DRUMS instrument (from disk #AD-007 which came with your ASR–10).

- Load STEREO DRUMS into an ***Instrument•Sequence Track*** and select it (its LED should be lit yellow).
- Press the ***Edit*** button, then ***System•MIDI***, and using the ***Arrow*** buttons, scroll to the ENTER PLAYS KEY parameter. The display shows ENTER PLAYS KEY=C4.
- Press ***Enter•Yes***. You will hear the ride cymbal assigned to C4, played at a velocity of 127. The note will sustain for as long as you hold down the ***Enter•Yes*** button. You can even press the Sustain pedal to sustain the note indefinitely. This note is transmitted via MIDI (assuming Edit/Instrument MIDI STATUS=BOTH) and can be recorded by the sequencer.
- Press the ***Up Arrow*** until the display shows ENTER PLAYS KEY=D4+.
- Press ***Enter•Yes***. You will hear now the crash cymbal assigned to D4+.
- Press the ***Up Arrow*** until the display shows ENTER PLAYS KEY=A4.
- Press ***Enter•Yes***. You will now hear the low tom assigned to A4.

Take a moment to select different keys using the ***Up/Down Arrow*** buttons, and play them using the ***Enter•Yes*** button. Fun, huh?

Now we're going to use the ENTER PLAYS KEY parameter to locate a WaveSample for editing. This is useful for determining which WaveSample to edit, without having a MIDI Controller connected to your ASR–10.

- Press the ***Up Arrow*** until the display shows ENTER PLAYS KEY=D3+.
- Press ***Enter•Yes***. You will hear the snare assigned to D3+.
- Press the ***Edit*** button. This takes us to the Edit Context page.
- Press the ***Enter•Yes*** button. This plays the snare again, and updates the Edit Context page (just as if a note-on for D3+ was received from a MIDI controller), showing which WaveSample (in Layer 1) is selected for editing. In this scenario, it is WS=6.

What if the WaveSample you want to edit is not in Layer 1 (or in the currently displayed layer)? By continually moving the cursor back and forth between the Layer and WaveSample values on the Edit Context page (selecting each defined layer), pressing the ***Enter•Yes*** button will help you locate the WaveSample that you want to edit.

- Press ***System•MIDI*** to go back to the ENTER PLAYS KEY parameter.
- Press the ***Up Arrow*** until the display shows ENTER PLAYS KEY=E6.
- Press ***Enter•Yes***. You will hear the hand clap assigned to E6.
- Press the ***Edit*** button to get to the Edit Context page.
- Press ***Enter•Yes*** to update the Edit Context page. You will hear the hand clap assigned to E6, but since this WaveSample is not on Layer 1, the display shows WS=ALL.
- Move the cursor under the Layer value, and select Layer 2 using the ***Up Arrow*** button.
- Move the cursor under the WaveSample value, and press the ***Enter•Yes*** button. You have just determined that this WaveSample is not on Layer 2, because the display still shows WS=ALL.
- Repeat the last to steps for Layer 3, Layer 4 and Layer 5. Haven't found it yet, huh?
- Move the cursor under the Layer value, and select Layer 6 using the ***Up Arrow*** button.
- Move the cursor under the WaveSample value, and press the ***Enter•Yes*** button. The WaveSample value has been updated, and the display shows WS=37. You've just used the ***Enter•Yes*** button to find the location of the hand clap (LYR=6 WS=37). Now press the ***Enter•Yes*** button repeatedly, and give yourself a hand!

Sequencing MIDI Applications

Section 17 — Sequencing/MIDI Applications

This section covers a number of advanced sequencer applications, including using the ASR–10 with a variety of external MIDI devices.

Using the ASR–10 as a Master MIDI Controller

The MIDI Connection

Musical Instrument Digital Interface (MIDI) — that magical connection that lets you play one instrument (or a whole roomful of them) from another — is a standard that has been agreed upon by manufacturers for translating musical events into specific numbers that are transmitted and received by MIDI instruments.

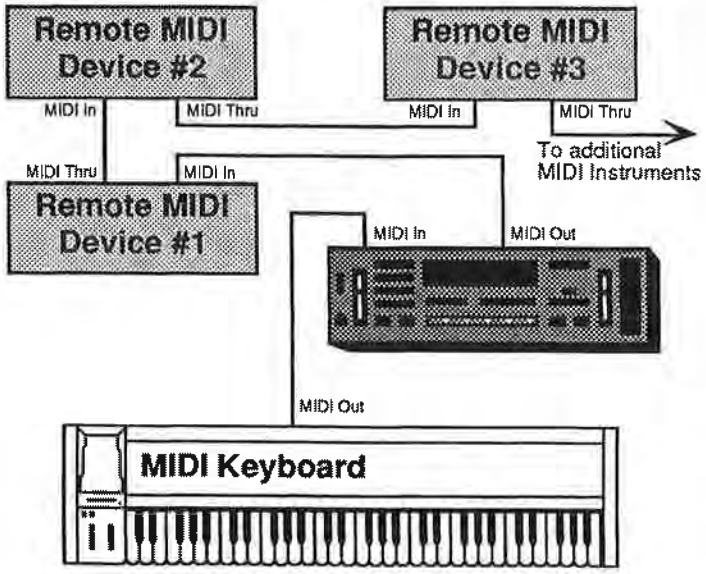
For instance, when you play middle C on theMIDI keyboard connected to the ASR–10, it instantly transmits a series of numbers out its MIDI Out jack. These numbers represent a *Key Down* event, along with the location of the note on the keyboard and how hard the key was struck. When you release the key, it sends the ASR–10 a number meaning *Key Up*. The ASR–10 can receive and translate those numbers and will play the middle C itself. The same thing happens whenever you move a controller, such as a pitch bend or mod wheel, or when you select a new sound — each of these events is translated into a series of numbers which are transmitted out the MIDI Out jack.

Controlling Remote MIDI devices from the ASR–10 — MIDI Connections

You can use the ASR–10 banks or performance presets to drive external MIDI instruments, greatly enhancing the number of available voices and timbres. With the Edit/Instrument MIDI STATUS parameter, an instrument, and its associated sequence and song track can be assigned one of the following settings:

- EXT or MIDI status — so that it will play only out via MIDI
- LOCAL status — so that it will play only locally on the ASR–10
- BOTH — in which case it will play a local sound *and* will transmit on its designated MIDI OUT CHANNEL

When controlling multiple remote MIDI devices, first connect the various devices to the ASR–10, and to each other, as shown here. Connect the MIDI Out jack of your MIDI Controller to the MIDI In jack of the ASR–10. Connect the MIDI Out jack of the ASR–10 to the MIDI In jack of the first device. Then connect the MIDI Thru jack of the first device to the MIDI In jack of the second device. Connect the MIDI Thru jack of the second device to the MIDI In jack of the third device. And so on, for as many devices as you will be using.



With this arrangement, once you set up the proper MIDI channels, etc., each remote MIDI device will receive and play only the data that is intended for it, and will "pass along" all other data. Also, each can be played from its own keyboard (as well as from the ASR–10's) without affecting the others, because MIDI Thru jacks only pass along incoming MIDI data, and do not transmit what is played on the instrument.

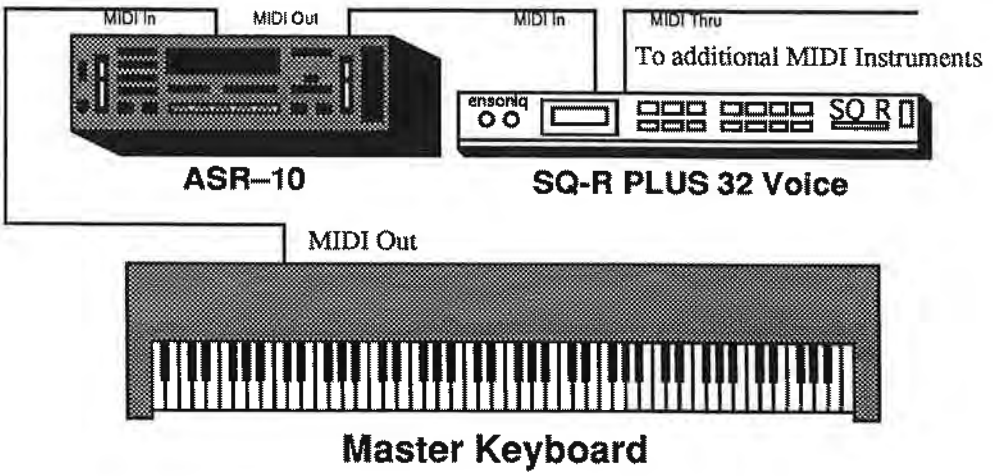
This set up is ideal for controlling everything right from the ASR–10. Simply by making a MIDI Instrument and setting the ***Instrument•Sequence Track*** to the same MIDI channel as a particular instrument, you can:

- Play the ASR–10 and other remote MIDI devices from the MIDI keyboard
- Record a sequence track that will play back on the remote MIDI device when you play the sequence or song
- Send the remote MIDI devices program changes and adjust their volume (assuming the devices receive MIDI volume)

In other words, once you have made the appropriate connections and set up the MIDI configuration of the tracks and all remote MIDI devices, you can use your main keyboard and the ASR–10's front panel to control and record all the remote MIDI devices in your rig.

Creating a MIDI Instrument

Let's suppose that you have a synthesizer module (say an ENSONIQ SQ–R PLUS 32 Voice) connected to the ASR–10 as shown below:



You want to be able to select the SQ–R PLUS 32 Voice from the ASR–10 front panel and then play or sequence it from your master controller just as you would an ASR–10 instrument. To do this, you can make an ASR–10 instrument that contains no sample data, and that will not sound on the ASR–10, but will only transmit via MIDI. Such "MIDI Instruments," as we call them, use up very little memory, and can be loaded from disk almost instantly. They are the key to using the ASR–10 as a MIDI master controller.

- On the Edit/System•MIDI page, set the TRANSMIT ON parameter to TRANSMIT ON= INST CHAN.
- Press ***Command***, then press ***Instrument***.
- Scroll left or right until the display reads CREATE NEW INSTRUMENT.
- Press ***Enter•Yes***. The display reads SELECT UNUSED INST= 1. The new instrument will be created in the ***Instrument•Sequence Track*** location indicated. You can select a different location (by pressing any unused ***Instrument•Sequence Track*** button), or just press ***Enter•Yes***.
- Press ***Enter•Yes***. The ASR–10 will create a new unnamed instrument in the location you indicated. The new instrument is selected (its LED is lit yellow). Note that it contains no layers or WaveSamples and will make no sound when you play the connected MIDI keyboard.
- Press ***Edit***, then press ***Instrument***. This puts you on the Edit/Instrument page where you will assign the characteristics of your new instrument.
- Using the ***Arrow*** buttons, scroll left (or right) until the display shows MIDI STATUS. Set MIDI STATUS = MIDI. Now the instrument will only play over MIDI, sending keys, controllers, etc. on its designated MIDI channel.
- Scroll left to the parameter MIDI OUT CHANNEL = __. This is the MIDI channel on which the instrument will transmit. Let's suppose that the SQ–R PLUS 32 Voice in our example is set to receive on MIDI channel 3. We would set this parameter to MIDI OUT CHANNEL= 3. Now, selecting and playing this instrument should play the SQ–R PLUS 32 Voice.

- Name the new instrument. Scroll left until the display shows NAME = UNNAMED INST and use the ***Arrow*** buttons to name/rename the instrument. Give the new instrument a name which describes it. In this case we might name it SQR - CHAN 3.

- Press ***Load*** to return to LOAD mode and select the MIDI instrument. The instrument's name and volume level show on the display like any other ASR–10 instrument. When you play it from the controller keyboard you should hear the SQ–R PLUS 32 Voice.
- You can select, deselect, or stack the MIDI instrument as you would a local instrument.
- You can change its keyboard range so it only plays over a specified part of the keyboard.
- Changing its volume (by moving the ***Data Entry Slider***) will send MIDI volume changes to the SQ–R PLUS 32 Voice, letting you mix its level just as you would a local instrument.
- Save the new instrument to disk. You can later load it into any of the eight ***Instrument•Sequence Track*** locations and play the SQ–R PLUS 32 Voice just as you would load and play a local sampled instrument.

Create customized MIDI-only ASR–10 instruments for each of your remote MIDI devices. Then load them into memory for playing and sequencing specific external instruments. Just remember to follow these rules:

- TRANSMIT ON = must be set to INST CHAN on the Edit/System•MIDI page.
- Each external device for which you create a MIDI instrument should always be set to receive in POLY (or MULTI) mode, OMNI OFF, and always on the same MIDI channel that you assign the related ASR–10 MIDI instrument to transmit on.
- Make sure that any ASR–10 instruments that you don't want transmitting notes via MIDI are assigned to SEND KEYS TO = LOCAL. Then save the instrument(s) to disk.

Recording into the ASR–10 Sequencer from a MIDI Controller

On many occasions, you may find it convenient to use an external controller to record a track into the ASR–10's internal sequencer. This could be for added authenticity (recording from Drum Pads to get a "live drummer" feel), performance flexibility (recording from a weighted-action MIDI keyboard controller like the ENSONIQ KS–32), or ease of use (a guitarist recording from a guitar controller). No matter which type of MIDI controller you use, there are a few parameters that need to be set in order to record into the sequencer.

Recording Onto a Single Track (One MIDI Channel)

- Connect a MIDI cable from the MIDI Out of the controller to the MIDI In of the ASR–10.
- Set the MIDI IN MODE to POLY on the Edit/System•MIDI page. The MIDI controller should be in POLY mode as well.
- Set the ASR–10 MIDI BASE CHANNEL to the same MIDI channel that the MIDI controller is set to transmit on.
- Select the track you want to record on by pressing the appropriate ***Instrument•Sequence Track*** button on the ASR–10. Make sure the Instrument on the selected ***Instrument•Sequence Track*** has its Edit/Instrument MIDI STATUS set to BOTH, LOCAL, or MIDI.
- On the Edit/Seq•Song page, set the SEQ REC SOURCE parameter to BOTH or MIDI.

Recording proceeds normally from this point, with the sequencer responding to the data from the MIDI controller.

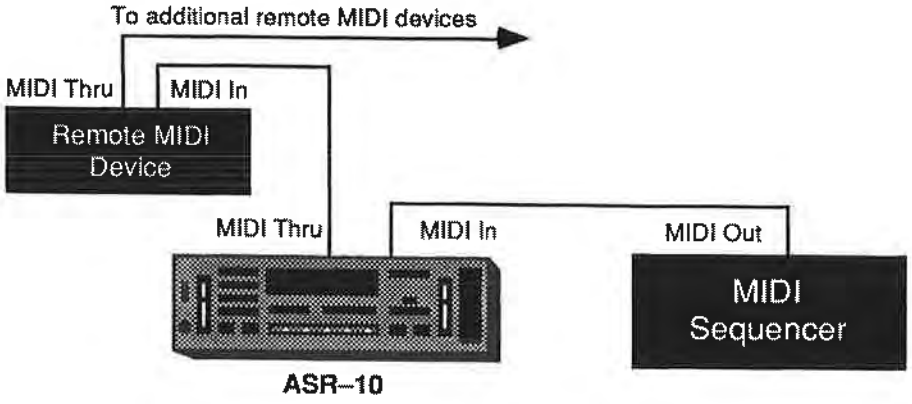
Recording Onto Several Tracks (Multiple MIDI Channels) — Multi-Track Record

- Connect a MIDI cable from the MIDI Out of the controller to the MIDI In of the ASR–10.
- Set the MIDI IN MODE to MULTI on the Edit/System•MIDI page.
- Set the MIDI controller to either MONO B (for guitar controllers sending on individual strings) or MULTI (for transferring data from another sequencer). Consult the manual for the external MIDI device for details.
- Set the individual MULTI IN MIDI CHANNEL for each track (found on the Edit/Track page), matching MIDI receive channels with those that will be transmitted on by the controller. Different MIDI receive channels must be selected for each ***Instrument•Sequence Track*** that you want to receive via MIDI. If more than one sequence track is set to the same MULTI IN MIDI CHAN, only the lowest numbered ***Instrument•Sequence Track*** will receive via MIDI.
- Make sure all of the instruments on the ***Instrument•Sequence Tracks*** that you want to record on have their Edit/Instrument MIDI STATUS set to BOTH, LOCAL, or MIDI.
- On the Edit/Seq•Song page, set the SEQ REC SOURCE parameter to MULTI.

Once these parameters are set, you can begin recording. For more information on recording with the SEQ REC SOURCE set to MULTI (Multi-Track Record), see *Section 12 — Sequencer and Audio Track Concepts*.

Using the ASR–10 with an External Sequencer

Because of its multi-timbral capabilities, the ASR–10 is an ideal instrument for use with a MIDI sequencer. Its ability to receive on up to eight MIDI channels at once means that the ASR–10 can take the place of several instruments in your rig.



The illustration above shows a typical sequencing set-up. Of course the ASR–10 doesn't have to be the first in the chain after the sequencer — you can chain MIDI devices in any order, with one exception; if a unit doesn't have a MIDI Thru jack, you must place that unit last in the chain.

POLY Mode — The ASR–10 as One Instrument

We'll start with the most basic sequencing situation, in which the ASR–10 will be used as if it were an ordinary instrument. This is how it will act in POLY mode. In POLY mode, the ASR–10 will respond only to information received on the base MIDI channel, and will play only the sounds that are selected on the front panel.

- On the Edit/System•MIDI page, set the MIDI mode parameter to MIDI IN MODE=POLY.
- Also on the Edit/System•MIDI page, set the MIDI channel (MIDI BASE CHANNEL=___) to the MIDI channel you want the ASR–10 to receive on. In POLY mode, the ASR–10 will only respond to MIDI messages received on this MIDI channel.

MULTI Mode — The ASR–10 as Eight Independent Instruments

When you set the MIDI IN MODE= MULTI on the Edit/System•MIDI page, the ASR–10 becomes eight "virtual instruments," each receiving on its own MIDI channel, but all sharing the same pool of voices.

- On the Edit/System•MIDI page, set the MIDI IN MODE parameter to MULTI.
- For each instrument, set the MULTI IN MIDI CHANNEL parameter, found on the Edit/Track page. The MIDI channels you select for each track should correspond to the MIDI channels that the external MIDI sequencer will transmit on.

Some Important Points about MULTI Mode

- Each of the eight ***Instrument•Sequence Tracks*** is completely independent and polyphonic. The ASR–10's dynamic voice allocation means that at any moment, each track can play all of the ASR–10's voices, if it needs them. If all voices are in use and a track needs a voice, it will "steal" the voice from the oldest note playing.
- In MULTI mode it doesn't matter what instruments are selected or stacked on the front panel. Which instruments will play depends entirely on what MIDI channel(s) the data is received on.
- Different MIDI receive channels must be selected for each ***Instrument•Sequence Track*** that you want to receive via MIDI. If more than one sequence track is set to the same MULTI IN MIDI CHAN, only the lowest numbered ***Instrument•Sequence Track*** will receive via MIDI.
- In MULTI mode, program changes received by a given ***Instrument•Sequence Track*** on its MIDI channel will cause the ASR–10 to try to load the same numbered instrument disk file into that ***Instrument•Sequence Track*** location, replacing the instrument that is currently loaded in that location.
- When using a MIDI sequencer, it's generally a good practice to assign a certain MIDI channel (or channels) to each instrument in your rig and always leave them set to that channel.
- When recording in MULTI mode, the SEQ REC SOURCE parameter on the Edit/Seq•Song page must also be set to MULTI.

Synchronizing to an External MIDI Clock Source

When you set the CLOCK SOURCE= MIDI on the Edit/Seq•Song page, the ASR–10 will sync to incoming MIDI clocks from a remote device. The MIDI indicator light will flash and the display will show WAITING... Recording will begin with the first MIDI clock received after a MIDI Start command. If the ***Stop•Continue*** button is pressed *before* a MIDI Start command is received, the ASR–10 waits 5 seconds before stopping.

Using the ASR–10 with a Drum Machine

When you use the ASR–10 in conjunction with a drum machine or other rhythm sequencer, there are three ways to have them operate together:

- Sync the drum machine's clock to the ASR–10
- Sync the ASR–10's clock to the drum machine
- Sequence the drum machine from the ASR–10, just as you would a synthesizer

To Sync a Drum Machine to the ASR–10:

- Connect the MIDI Out of the ASR–10 to the MIDI In of the drum machine.
- Set the drum machine to sync to MIDI clocks.
- Set the drum machine to receive on an unused MIDI Channel, OMNI Off; or disable channel information. You don't want the drum machine receiving and playing back ASR–10 sequence data intended for other instruments. MIDI clocks, start, stop, and continue are *real time* commands that are sent and received regardless of MIDI channel or mode.
- The drum machine should now sync to the ASR–10's clock. Pressing ***Play*** or ***Stop•Continue*** will start, stop, and continue the drum machine, assuming it receives those commands.

To Sync the ASR–10 to a Drum Machine:

- Connect the MIDI Out of the drum machine to the MIDI In of the ASR–10.
- Set the ASR–10 to sync to MIDI clocks. On the Edit/Seq•Song page, select CLOCK SOURCE=MIDI.
- Set the drum machine to not transmit channel information, or to transmit on a MIDI Channel that none of the ASR–10 ***Instrument•Sequence Tracks*** have their Edit/Track MULTI-IN MIDI CHANNEL set to receive on. Again, MIDI clocks, start, stop, and continue are sent and received regardless of MIDI channel or mode.
- The ASR–10 should now sync to the drum machine's clock. Starting, stopping, or continuing the drum machine will start, stop, and continue the ASR–10.

To Sequence a Drum Machine from a Track of the ASR–10:

- Connect the MIDI Out of the ASR–10 to the MIDI In of the drum machine.
- Set the drum machine to Tape Sync or External Clock, or any setting other than Internal or MIDI Clock. This way it will not play its own patterns, but will act only as a sound-producing device, sequenced from a track of the ASR–10.
- Set the drum machine to POLY (OMNI Off) mode and select a MIDI channel.
- From the Edit/Instrument page, set the MIDI Status for an ASR–10 Instrument to MIDI. Set the instrument's MIDI OUT CHANNEL to the same MIDI channel you assigned the drum machine to receive on.
- You should now be able to play the drum machine from the ASR–10 keyboard. You can then record a track on the ASR–10, from the ASR–10 keyboard, which will play on the drum machine — just as if you were sequencing an external synthesizer. The advantage of this approach is that some drum machines respond better to velocity when played via MIDI than when played from their own front panels. Thus, you may get more dynamic range out of your drum machine if you use this approach. The disadvantage is that you use up ASR–10 sequencer memory to sequence the drum machine.

Using the ASR–10 with a MIDI Guitar Controller

The ASR–10 is an ideal instrument to use with any MIDI guitar controller that is capable of transmitting in MONO mode. MONO mode allows a guitar controller to transmit the notes played on each string on a different MIDI channel. This has the advantage of letting each string send pitch bends independently, which is the only way to truly recreate guitar technique on a synthesizer.

Some earlier guitar controllers do not support MONO mode. You will have to consult the manual of your particular model to see if it does. If you have a guitar controller that only sends in POLY mode (i.e. send all six strings on the same MIDI channel) you should use the ASR–10 in POLY mode or OMNI mode.

For MIDI guitar controllers that do support MONO mode, you will want to put the ASR–10 in one of the two types of MONO mode that are available. In MONO A, the ASR–10 receives on eight consecutive MIDI channels with only one voice assigned to each MIDI channel. In MONO B, the ASR–10 receives on the Edit/Track MULTI-IN MIDI channel assigned to each ***Instrument•Sequence Track***.

- MONO A is the mode you will probably use most often. This is the mode to use when you want to play the same sound on all the strings of your guitar controller. In MONO A, the ASR–10 receives on eight consecutive MIDI channels (the MIDI BASE CHANNEL through MIDI BASE CHANNEL +7) and will play whatever ***Instrument•Sequence Track***(s) are selected or stacked on the front panel. In other words, the ASR–10 behaves as it does in POLY or OMNI modes, except that it receives monophonically on multiple MIDI channels. This provides the advantage of multiple tracks that will respond independently to controllers received on multiple channels, but you do not have to set up the instruments for each track separately.
- MONO B is the mode to use if you want to be able to play a different sound on each string of your guitar controller. In MONO B, each ***Instrument•Sequence Track*** receives on its own Edit/Track MULTI-IN MIDI channel. In other words, the ASR–10 behaves as it does in MULTI mode, except that each ***Instrument•Sequence Track*** receives monophonically on multiple MIDI channels. This is the only way to get a different sound on each string when using a MIDI guitar controller.

Using MONO A Mode

To use the ASR–10 in MONO A mode, set it up as follows:

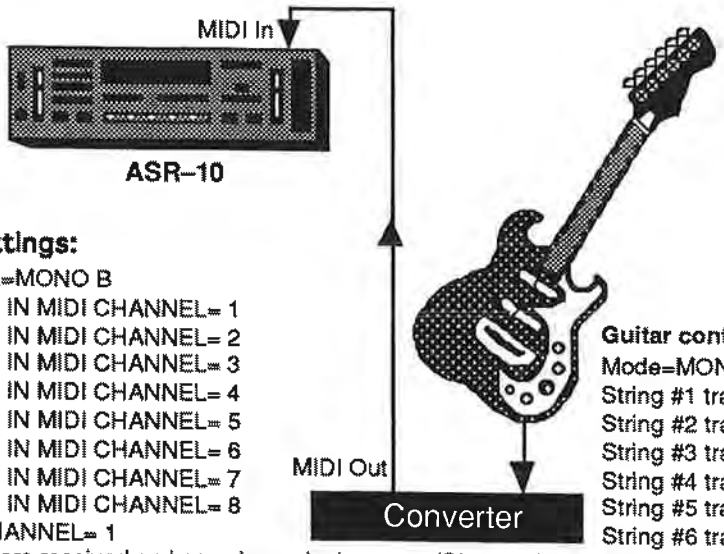
- Load one or more instruments into the ASR–10.
- Press ***Edit***, then ***System•MIDI***. Scroll to MIDI IN MODE.
- Select MIDI IN MODE= MONO A.
- Set your guitar controller to transmit in MONO mode on channels 1– 6 (some models have an easy shortcut for getting into this state).
- Connect the MIDI Out of the guitar controller module to the MIDI In of the ASR–10.
- Press ***Load*** on the ASR–10 to return to Load mode (remember — the ASR–10 will only respond to program changes or select multiple instruments when in Load mode).

- Select an ***Instrument•Sequence Track*** either by pressing a front panel button or by sending a program change from the guitar controller.
- You should now be able to play the ASR–10 from the guitar controller with all six strings playing whatever instrument(s) are selected or stacked on the front panel.

Using MONO B Mode

To use the ASR–10 in MONO B mode, set it up as follows:

- Load one or more instruments into the ASR–10.
- Press ***Edit***, then ***System•MIDI***. Scroll to MIDI IN MODE.
- Select MIDI IN MODE= MONO B.
- Press ***Edit***, then ***Track***. Scroll to MULTI-IN MIDI CHANNEL.
- Set each ***Instrument•Sequence Track*** MULTI-IN MIDI CHANNEL to correspond with its track number (Track 1 MULTI-IN MIDI CHANNEL=1, Track 2 MULTI-IN MIDI CHANNEL=2, etc.).
- Set your guitar controller to transmit in MONO mode on channels 1– 6 (some models have an easy shortcut for getting into this state).
- Connect the MIDI Out of the guitar controller module to the MIDI In of the ASR–10.
- You should now be able to play the ASR–10 from the guitar controller with each string playing its corresponding ***Instrument•Sequence Track*** as shown below:



ASR–10 Settings:
MIDI IN MODE=MONO B
Track 1 MULTI IN MIDI CHANNEL= 1
Track 2 MULTI IN MIDI CHANNEL= 2
Track 3 MULTI IN MIDI CHANNEL= 3
Track 4 MULTI IN MIDI CHANNEL= 4
Track 5 MULTI IN MIDI CHANNEL= 5
Track 6 MULTI IN MIDI CHANNEL= 6
Track 7 MULTI IN MIDI CHANNEL= 7
Track 8 MULTI IN MIDI CHANNEL= 8
MIDI BASE CHANNEL= 1
Global Controllers received on base channel minus one (Chan. 16)

Guitar controller Settings
Mode=MONO
String #1 transmits on Channel 1
String #2 transmits on Channel 2
String #3 transmits on Channel 3
String #4 transmits on Channel 4
String #5 transmits on Channel 5
String #6 transmits on Channel 6
Global controllers transmitted on Chan. 16

A few important points about MONO B Mode:

- Notes played on each string will play only the corresponding ***Instrument•Sequence Track***. Each string/***Instrument•Sequence Track*** combination is totally independent.
- MULTI-IN MIDI CHANNELs do not have to be set 1-8. They can be set to any channel from 1 to 16, as long as each track has a different MULTI-IN MIDI CHANNEL and the guitar controller transmits accordingly. Unless you have a very complex MIDI setup, setting things up as shown above will provide the most straight-forward and intuitive arrangement.
- Different MIDI receive channels must be selected for each ***Instrument•Sequence Track*** that you want to receive via MIDI. If more than one sequence track is set to the same MULTI IN MIDI CHAN, only the lowest numbered ***Instrument•Sequence Track*** will receive via MIDI.

Global Controllers in MONO A and B Modes

Global controllers are controllers sent on one channel that affect all other channels simultaneously. They can be useful in reducing the number of MIDI events required to achieve particular effects, and can thereby reduce the delays sometimes associated with overloading MIDI. Some guitar controllers can transmit global controllers, and the ASR–10 can respond to them.

In MONO mode (A or B) the base channel minus one becomes the MIDI channel for global controllers (pitch bend, pressure, etc.). For example, if the base channel is channel 3, any controllers received on channel 2 will be interpreted as global controllers and will affect *all* voices being played. If the base channel is channel 1, channel 16 becomes the channel for global controllers. Each track will also respond independently to controllers sent on its own channel. For example, each guitar string on a MIDI guitar can send independent pitch bend, while the "whammy bar" controller could be sent on the global channel (channel 16 in the example above) to affect all voices.

Note: The ASR–10 will not receive note data via MIDI on the base channel minus one in MONO A and B modes. Therefore, we recommend that the MIDI BASE CHANNEL parameter on the Edit/System•MIDI page be set to the same channel as the ***Instrument•Sequence Track*** with the lowest numbered Edit/Track MULTI-IN MIDI CHAN number.

Patch Selects and Presets

Making patch select changes from a continuous controller (i.e. mod wheel, whammy bar, etc.) can be an adventure. The patch select settings are sent and received as MIDI controller #70. Since the ASR–10 recognizes the 0–31 range as both patch selects up; 32–63 as the right patch select down; 64–95 as the left patch select; 96–127 as both patch selects down, it can be difficult to make predictable changes from a continuous controller. If your MIDI guitar controller has the capability of sending discrete or specific controller values, rather than continuous controller numbers, then you'll be able to send predictable patch select changes.

Another alternative is to use the two pedals of the optional SW-10 Dual Foot Switch. This way you can change patches in real time with your feet as you play the MIDI guitar controller.

Also, patch select configurations can be stored within performance presets which can be accessed by program changes 17-24 (in MONO A mode only). Each time you send a program change from your MIDI guitar controller, the ASR–10 will respond to that message by selecting one of the eight presets. Check the program change and patch assignment section in your MIDI guitar controller manual.

Song Position Pointers

The ASR–10 sends and receives Song Position Pointers via MIDI. Song Position Pointers are MIDI commands that tell a sequencer or drum machine where to locate within a song or sequence.

When the ASR–10 receives a Song Position Pointer, it will locate to the appropriate place in the selected song or sequence.

The ASR–10 sends a Song Position Pointer over MIDI whenever you use the Auto-Locate control (the GOTO function on the Edit/Seq•Song page). Any receiving unit which recognizes Song Position Pointers will locate to the same spot. (Not all devices recognize Song Position Pointers. Consult the manual of any other sequencing device you are using to see if it does.)

MIDI Song Selects

MIDI Song Selects allow a sequencer such as the ASR–10 to instruct a remote sequencer or drum machine to select a new song. The song can be assigned a MIDI Song Select number from 0 through 127. MIDI Song Select numbers 0 through 127 are transmitted via MIDI when you load a new song into the ASR–10, or when the currently loaded song is selected. The MIDI SONG SELECT parameter (on the Edit/System•MIDI page) must be set to "ON" for this to work.

The SEQUENCER INFORMATION parameter on the Command/Seq•Song page allows you to set the MIDI SONG SELECT number that will be sent whenever the song is selected. MIDI Song Select numbers are stored with the song when you save the song to disk.

Delay Times/Tempo BPM Chart

This chart shows the relationship between delay times and tempo beats per minute. You can use this chart to set the effect delay times to sync to your sequence or song.

Delay Times/Tempo BPM Chart

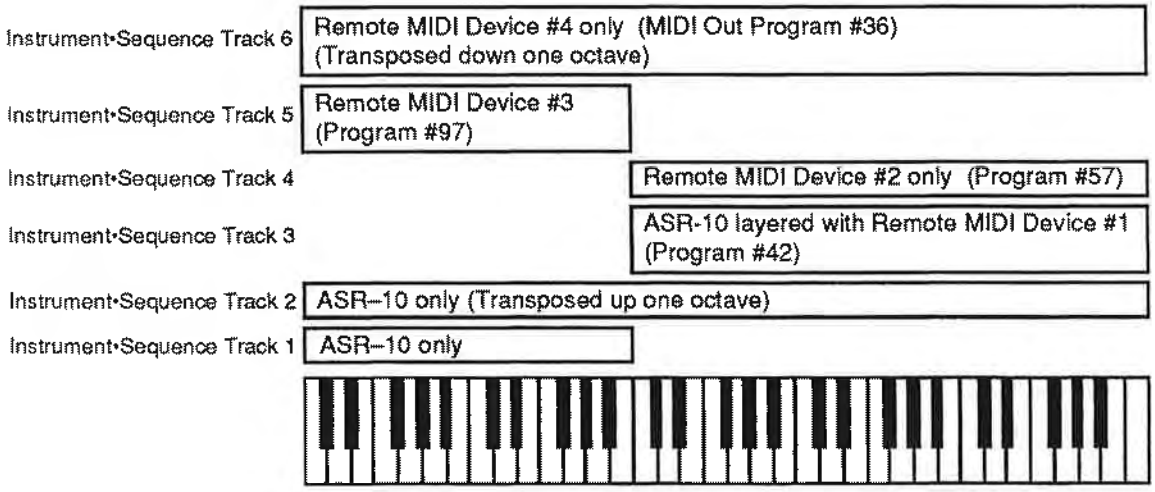
Delay Time	BPM	Delay Time	BPM	Delay Time	BPM	Delay Time	BPM	Delay Time	BPM	Delay Time	BPM
3.000	20	1.017	59	0.612	98	0.438	137	0.341	176	0.279	215
2.857	21	1.000	60	0.606	99	0.435	138	0.339	177	0.278	216
2.727	22	0.984	61	0.600	100	0.432	139	0.337	178	0.276	217
2.609	23	0.968	62	0.594	101	0.429	140	0.335	179	0.275	218
2.500	24	0.952	63	0.588	102	0.426	141	0.333	180	0.274	219
2.400	25	0.938	64	0.583	103	0.423	142	0.331	181	0.273	220
2.308	26	0.923	65	0.577	104	0.420	143	0.330	182	0.271	221
2.222	27	0.909	66	0.571	105	0.417	144	0.328	183	0.270	222
2.143	28	0.896	67	0.566	106	0.414	145	0.326	184	0.269	223
2.069	29	0.882	68	0.561	107	0.411	146	0.324	185	0.268	224
2.000	30	0.870	69	0.556	108	0.408	147	0.323	186	0.267	225
1.935	31	0.857	70	0.550	109	0.405	148	0.321	187	0.265	226
1.875	32	0.845	71	0.545	110	0.403	149	0.319	188	0.264	227
1.818	33	0.833	72	0.541	111	0.400	150	0.317	189	0.263	228
1.765	34	0.822	73	0.536	112	0.397	151	0.316	190	0.262	229
1.714	35	0.811	74	0.531	113	0.395	152	0.314	191	0.261	230
1.667	36	0.800	75	0.526	114	0.392	153	0.313	192	0.260	231
1.622	37	0.789	76	0.522	115	0.390	154	0.311	193	0.259	232
1.579	38	0.779	77	0.517	116	0.387	155	0.309	194	0.258	233
1.538	39	0.769	78	0.513	117	0.385	156	0.308	195	0.256	234
1.500	40	0.759	79	0.508	118	0.382	157	0.306	196	0.255	235
1.463	41	0.750	80	0.504	119	0.380	158	0.305	197	0.254	236
1.429	42	0.741	81	0.500	120	0.377	159	0.303	198	0.253	237
1.395	43	0.732	82	0.496	121	0.375	160	0.302	199	0.252	238
1.364	44	0.723	83	0.492	122	0.373	161	0.300	200	0.251	239
1.333	45	0.714	84	0.488	123	0.370	162	0.299	201	0.250	240
1.304	46	0.706	85	0.484	124	0.368	163	0.297	202	0.249	241
1.277	47	0.698	86	0.480	125	0.366	164	0.296	203	0.248	242
1.250	48	0.690	87	0.476	126	0.364	165	0.294	204	0.247	243
1.224	49	0.682	88	0.472	127	0.361	166	0.293	205	0.246	244
1.200	50	0.674	89	0.469	128	0.359	167	0.291	206	0.245	245
1.176	51	0.667	90	0.465	129	0.357	168	0.290	207	0.244	246
1.154	52	0.659	91	0.462	130	0.355	169	0.288	208	0.243	247
1.132	53	0.652	92	0.458	131	0.353	170	0.287	209	0.242	248
1.111	54	0.645	93	0.455	132	0.351	171	0.286	210	0.241	249
1.091	55	0.638	94	0.451	133	0.349	172	0.284	211	0.240	250
1.071	56	0.632	95	0.448	134	0.347	173	0.283	212		
1.053	57	0.625	96	0.444	135	0.345	174	0.282	213		
1.034	58	0.619	97	0.441	136	0.343	175	0.280	214		

Hint: Assign a controller to modulate the effect delay times, and select a specified minimum and maximum range so that you can create unique poly-rhythms in real time.

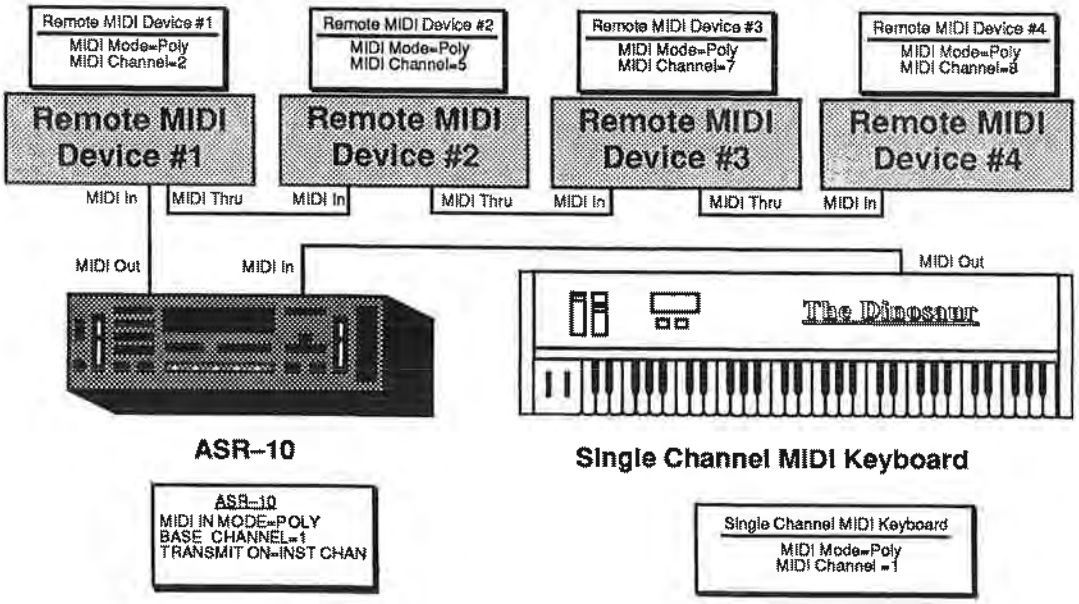
The Smart Transmit™ Function

The ASR–10 Smart Transmit Function is a unique and powerful feature that allows the ASR–10 to add new MIDI master controller features to any keyboard hooked up to it. Connect your favorite keyboard and it becomes an eight zone controller with independent control over volume, transpose, key range, MIDI channel, and program change remapping. The major advantage of the Smart Transmit function is that it gives you the power and flexibility of a controller keyboard, even if your MIDI keyboard lacks these multi-channel abilities. Smart Transmit also allows you to hear additional MIDI instruments that you are sequencing with the ASR–10 internal sequencer without the hassle of resetting MIDI parameters on your controller keyboard each time you switch tracks.

Let's take a look at how to implement and utilize the Smart Transmit function. First, we'll need to determine which local ASR–10 instruments and which Remote MIDI Devices should play across specific areas of the keyboard. For example:


Instrument•Sequence Track 6
Remote MIDI Device #4 only (MIDI Out Program #36)
(Transposed down one octave)
Instrument•Sequence Track 5
Remote MIDI Device #3
(Program #97)
Instrument•Sequence Track 4
Remote MIDI Device #2 only (Program #57)
Instrument•Sequence Track 3
ASR-10 layered with Remote MIDI Device #1
(Program #42)
Instrument•Sequence Track 2
ASR–10 only (Transposed up one octave)
Instrument•Sequence Track 1
ASR–10 only

The next step is to make MIDI connections between all the devices and set their system controls to receive incoming MIDI. The diagram below illustrates the connections and MIDI settings:



With the MIDI connection made and the parameters set properly, the final step is to select the proper settings on the Edit/Instrument page.

The chart below shows the settings that would be used to create the multi-timbral split/layer combination in this example. Refer to the descriptions in *Section 4 — Instrument, Bank, and Preset Parameters* for detailed information on each of the parameters.

Track	1	2	3	4	5	6	7	8
Goal	Play ASR Sound	Play ASR Sound	ASR–10 MIDI Ch. 2	Play MIDI Channel 5	Play MIDI Channel 7	Play MIDI Channel 8	OFF	OFF
MIDI OUT CHANNEL	N/A	N/A	2	5	7	8	N/A	N/A
MIDI OUT PROGRAM	N/A	N/A	042	057	097	036	N/A	N/A
MIDI STATUS	LOCAL	LOCAL	BOTH	EXT	EXT	EXT	N/A	N/A
INST (Key) RANGE	C2-B3	C2-C7	C4-C7	C4-C7	C2-B3	C2-C7	N/A	N/A
XPOS (Transpose)	+0 +00	+1 +00	+0 +00	+0 +00	+0 +00	-1 +00	N/A	N/A

Note: You must have TRANSMIT ON= INST CHAN and MIDI IN MODE=POLY on the Edit/System•MIDI page in order for Smart Transmit to work. Also remember to have the cables to any Remote MIDI Devices connected to the ASR–10's MIDI **Out** jack, *not* the MIDI Thru jack. The MIDI transmission channel of your MIDI controller must match the Edit/System•MIDI MIDI BASE CHANNEL parameter setting on your ASR–10.

After setting up these parameters, save the bank, giving it a name that describes the set-up. Once this is done, whenever you load the bank, each ***Instrument•Sequence Track*** with its Edit/Instrument MIDI STATUS set to BOTH, MIDI, or EXT will transmit the appropriate program changes, volumes, etc. to the receiving instruments on its MIDI OUT CHANNEL (with MIDI STATUS= LOCAL, ASR–10 instruments will be set up with the correct parameter settings as well).

Pressure

The ASR–10 Smart Transmit function also allows you to retransmit (and in some cases alter) the type of pressure an ***Instrument•Sequence Track*** receives. The chart below shows the relationship between type of pressure received, ***Instrument•Sequence Track*** pressure type (the PRESSURE MODE parameter setting on the Edit/Instrument page), and the pressure type retransmitted:

When the ASR–10 receives:	with PRESSURE MODE set to:	The ASR–10 will retransmit:
Channel	NONE	No Pressure
Channel	CHAN	Channel Pressure
Channel	KEY	No Pressure
Key	NONE	No Pressure
Key	CHAN	Channel Pressure
Key	KEY	Key Pressure

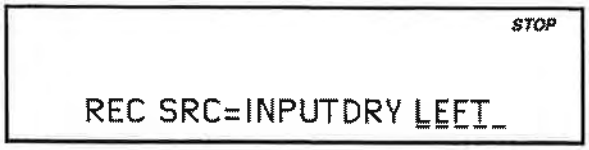
Audio Track Applications

Section 18 — Audio Track Applications

Using a Microphone with the ASR–10

Because the Audio Tracks allow continuous audio monitoring of the stereo Audio Inputs, you can sing along (through effects if desired) while playing the MIDI keyboard connected to the ASR–10. Here's how:

- Load in the Instruments that you want to use (to sing along with) by pressing ***Load***, then ***Instrument***, then scrolling through the files with the ***Up/Down Arrow*** buttons.
- Press ***Enter•Yes*** and the appropriate ***Instrument•Sequence Track*** button to load the currently viewed Instrument/Bank file into the internal memory. See *Section 1 — Controls and Architecture* for more information about loading files.
- When using a microphone with the ASR–10, we recommend plugging it into a mixer or preamp, and sending a line from the mixer/preamp to the Left Audio Input jack, for optimal performance.
- On the front panel of the ASR–10, press the ***Sample•Source Select*** button. The display shows the record source screen:



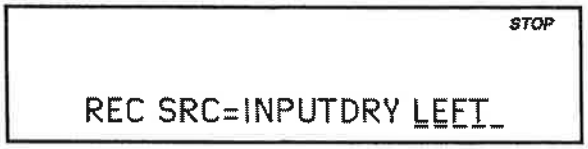
The record source value defaults to exactly what we want. You may have noticed when you pressed ***Sample•Source Select*** that the left Audio Track yellow LED turned on. When lit, it indicates that the track is selected.

- Press the left ***Audio Track*** button so that the red LED is lit. You should now be hearing the microphone signal though the ASR–10.
- Adjust the Input Level control (on the rear panel) so that the Signal is present (green Input Level LED lit) but only peaks occasionally (red Input Level LED lit). Remember, the red Input Level LED lights at 6dB below clipping.
- At this point, you can play different sounds (by pressing their ***Instrument•Sequence Track*** buttons) while singing along with the ASR–10.

Using a Guitar with a Sequence or Song

Using a guitar (or any line level signal) is almost identical to using a microphone, but with different input level settings. The Audio Tracks allow you to play along (through effects if desired) with the ASR–10 sequencer. Here's how:

- Load in the Instruments that you want to use by pressing ***Load***, then ***Instrument***, then scrolling through the files with the ***Up/Down Arrow*** buttons.
- Press ***Enter•Yes*** and the appropriate ***Instrument•Sequence Track*** button to load the currently viewed Instrument/Bank file into the internal memory.
- Load in the Sequences or Song that you want to use (to play along with) by pressing ***Load***, then ***Seq•Song***, then scrolling through the files with the ***Up/Down Arrow*** buttons.
- Press ***Enter•Yes*** to load the currently viewed Sequence/Song file into the internal memory. These may be single sequence files or a complete song file consisting of many sequences (a song may also be loaded as part of a bank file).
- Plug the guitar (or any line level signal) directly into the left Audio Input jack on the rear panel.
- On the front panel of the ASR–10, press the ***Sample•Source Select*** button. The display shows the record source screen:



The record source value defaults to exactly what we want. You may have noticed when you pressed ***Sample•Source Select*** that the left Audio Track yellow LED turned on. When lit, it indicates that the track is selected.

- Press the left ***Audio Track*** button (red LED lit). You should now be hearing the guitar (or other line level signal) signal though the ASR–10.
- Adjust the Input Level control (on the rear panel) so that the signal is present (green Input Level LED lit) but only peaks occasionally (red Input Level LED lit). Remember, the red Input Level LED lights at 6dB below clipping.
- At this point, you can play sequences or songs (by pressing the ***Play*** button) while playing your guitar (or other line level signal) along with the ASR–10.

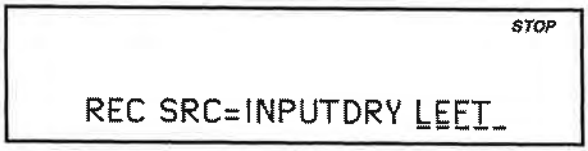
Assigning Different Output Routings

When using the Audio Tracks with an effect, you can assign different output routings for each Audio Track, as well as for each ***Instrument•Sequence Track***. For instance, lets say you have a sequence/song with a drum kit, bass, piano, and a horn section on ***Instrument•Sequence Tracks 1, 2, 3,*** and ***4*** respectively, and a microphone signal plugged into ***Audio Track A***. For this example, we'll be using FX=ROM-10 CHOR+REV+DDL. By assigning different outputs, you can route:

- the drum kit through reverb
- the bass through the chorus and reverb
- the piano through the delay line
- the horn section dry, without any effect processing
- the microphone through just the reverb

You can see that by assigning different output routings, you can create different effect variations. Here's how to set up the above scenario:

- Load a drum kit onto ***Instrument•Sequence Track 1.***
- Load a bass onto ***Instrument•Sequence Track 2.***
- Load a piano onto ***Instrument•Sequence Track 3.***
- Load a horn section onto ***Instrument•Sequence Track 4.***
- Load in any song or sequences that you may want to use (to play or sing along with) by pressing ***Load***, then ***Seq•Song***, then scrolling through the files with the ***Up/Down Arrow*** buttons.
- Press ***Enter•Yes*** to load the currently viewed Seq•Song file into the internal memory.
- When using a microphone, we recommend plugging it into a mixer or preamp, and sending a line from the mixer/preamp to the Left Audio Input jack, for optimal performance.
- On the front panel of the ASR–10, press the ***Sample•Source Select*** button. The display shows the record source screen:



- Press the left ***Audio Track*** button so that the red LED is lit. You should now be hearing the microphone signal though the ASR–10.
- Adjust the Input Level control (on the rear panel) so that the signal is present (green Input Level LED lit) but only peaks occasionally (red Input Level LED lit). Remember, the red Input Level LED lights at 6dB below clipping.
- Press the ***FX Select•FX Bypass*** button and use the ***Up/Down Arrow*** buttons to select FX=ROM-10 CHOR+REV+DDL. The display shows:

- Press ***Edit***, then ***Effects***. The display shows the Effect variations screen for the Chorus+Reverb+Delay effect. You can select different parameters for editing by scrolling with the ***Left/Right Arrow*** buttons, and changing the values with the ***Up/Down Arrow*** buttons. Remember, as soon as you change a value, or edit any parameter, the ROM effect becomes the Bank effect.

Once you've edited the effect for your particular application, you can assign different output routings (you can still go back and edit the effect parameters after you've assigned the output routings). To assign the output routings:

- Select ***Instrument•Sequence Track 1*** (the drum kit). The LED for ***Instrument•Sequence Track 1*** is solidly lit yellow.
- Press ***Edit***, then ***Track***. Using the ***Left/Right Arrow*** buttons, scroll until the display shows OUT= (output name). With the ***Up/Down Arrow*** buttons, select OUT= BUS2 JUST REVERB.
- Select ***Instrument•Sequence Track 2*** (the bass). You should still be on the Edit/Track OUT= screen. Notice that the ***Instrument•Sequence Track 2*** LED is solidly lit yellow.
- With the ***Up/Down Arrow*** buttons, select OUT= BUS1 CHORUS+REVRB.
- Select ***Instrument•Sequence Track 3*** (the piano). You should still be on the Edit/Track OUT= screen. Notice that the ***Instrument•Sequence Track 3*** LED is solidly lit yellow.
- With the ***Up/Down Arrow*** buttons, select OUT= BUS3 DELAY LINE.
- Select ***Instrument•Sequence Track 4*** (the horn section). You should still be on the Edit/Track OUT= screen. Notice that the ***Instrument•Sequence Track 4*** LED is solidly lit yellow.
- With the ***Up/Down Arrow*** buttons, select OUT= AUX 1. This will send the audio output of the Horn Section *only* to the AUX 1 Output jacks, and not to the Main Outputs. The output of the AUX jacks is always dry, and can be connected to an additional external effects processor, like the ENSONIQ DP/4, for instance.
- Select ***Audio Track A*** (the microphone). Although you have now selected an Audio Track, you should still see its Edit/Track OUT= screen. The ***Audio Track A*** yellow LED is solidly lit. Notice that the last selected ***Instrument•Sequence Track*** LED is flashing. This indicates that it can still be played from the keyboard, but is no longer selected for editing.
- With the ***Up/Down Arrow*** buttons, select OUT= BUS2 JUST REVERB.
- Press the ***Load*** button, to return to solid Load mode.

Now in solid Load mode, when you play the instruments, sing in the microphone, or play a song or sequence, you can hear how the effect routings affect each sound differently.

Controls and Architecture 1
System•MIDI 2
Instrument, Bank, and Preset Concepts 3
Instrument, Bank, and Preset Parameters 4
Effect Concepts 5
Effect Parameters 6
Sampling/Signal Source Concepts 7
Sampling/Signal Source Parameters 8
WaveSample and Layer Concepts 9
Wave Data Parameters 10
WaveSample and Layer Parameters 11
Sequencer and Audio Track Concepts 12
Sequencer and Audio Track Parameters 13
Storage 14
Sampling Applications 15
Instrument Programming Applications 16
Sequencing/MIDI Applications 17
Audio Track Applications 18

Controls and Architecture

System•MIDI

Instrument, Bank, and Preset Concepts

Instrument, Bank, and Preset Parameters

Effect Concepts

Effect Parameters

Sampling/Signal Source Concepts

Sampling/Signal Source Parameters

WaveSample and Layer Concepts

Wave Data Parameters

WaveSample and Layer Parameters

Sequencer and Audio Track Concepts

Sequencer and Audio Track Parameters

Storage

Sampling Applications

Instrument Programming Applications

Sequencing/MIDI Applications

Audio Track Applications

Appendix

Appendix

ASR–10 MIDI Implementation

The ASR–10 features an extensive MIDI (Musical Instrument Digital Interface) implementation. For normal applications, you will find all the information you need regarding the ASR–10's MIDI functions in this manual. You can also refer to the MIDI Implementation Chart on the next page for a summary of the ASR–10 MIDI implementation.

If you are writing a computer program to communicate with the ASR–10 via MIDI, or otherwise require a copy of the full ASR–10 System Exclusive Specification, it is available free of charge by writing to:

ENSONIQ Corp
MIDI Specification Desk
155 Great Valley Parkway
P.O. Box 3023
Malvern PA 19355
USA

Include in your written request your name and address, and indicate that you would like a copy of the "ASR–10 System Exclusive Specification." Please allow 2 to 3 weeks for delivery.

Registered Parameters

Registered parameters are transmitted (as four sequential continuous controller messages) by the ASR-10 whenever certain parameters are edited from the front panel. The two registered parameter controllers select the parameter and the following two data entry controllers specify the value.

Controllers

Number	*Name*	*Value*
100	Registered Parameter Select LSB	00 or 01
101	Registered Parameter Select MSB	always 0
6	Data Entry MSB (Most Significant Byte)	(see below)
38	Data Entry LSB(Least Significant Byte)	(see below)

Registered Parameters

Number	*Name*	*ASR-10 parameter range*
00	Pitch Bend Range	0..12
01	Master Tuning (Fine)	0..255 (displayed -99..+99)

The parameter values are sent as twoData Entry controller messages:

Parameter	*Data Entry MSB (6)*	*Data Entry LSB (38)*
Pitch Bend Range	0..12	0
Master Tune	0..127 (internal bits 1..7)	64 or 0 (internal bit 0)

For Master Tune, which is an 8-bit value internally, the most significant 7 bits are offset by 64 and shifted once before being sent as Data Entry MSB (Controller 6). The least significant bit of the internal value is transmitted as bit 6 of Data Entry LSB (Controller 38).

MODEL: ASR–10 **MIDI Implementation Chart** **Version: 1.0**

Function...		Transmitted*	Recognized	Remarks
Basic Channel	Default	1	1	
	Channels	1-16	1-16	
Mode	Default	1	1, 3, 4, MULTI	current mode is memorized
	Messages	X	X	
	Altered	X	X	Global Controllers in MONO Mode
Note Number	True Voice	21-108	21-108	
Velocity	Note ON	O	O	
	Note OFF	X [1]	X	[1] A Note Off velocity of 64 is always sent for all keys.
After Touch	Key	O	O	PolyKey pressure
	Channel	O	O	
Pitch Bender		O	O	
Control Change		1 Mod Wheel 4 Foot (Pedal) 6 Data Entry MSB[2] 7 Volume 38 Data Entry LSB[2] 64 Sustain 70 Sound Variation[3] 100 Reg. Param. Select LSB[4] 101 Reg. Param. Select MSB[5]	1–95 External Controller 1 Mod Wheel 4 Foot (Pedal) 6 Data Entry MSB[2] 7 Volume 38 Data Entry LSB[2] 64 Sustain 70 Sound Variation[3] 100 Reg. Param. Select LSB[4] 101 Reg. Param. Select MSB[5]	assignable (XCTRL) [2] for Reg. Params only after 100 & 101 are transmitted or received. [3] Patch Selects; values of 0, 32, 64, 127 [4] values of 0 & 1 only [5] always 0
Program Change	True Number	0-127	0-23 0-38, 100-127	in OMNI, POLY,& MONO A in MULTI & MONO B
System Exclusive		O	O	see ASR-10 SysEx Spec.
System Common	: Song Pos	O	O	
	: Song Sel	O	X	
	: Tune	X	X	
System Real Time	: Clocks	O	O	
	: Commands	O [6]	O [6]	[6] Start, Stop, Continue
Aux. Messages	: Local On/Off	X	X	
	: All Notes Off	X	O [7]	[7] recognized by rackmount units only.
	: Active Sense	X	X	
	: Reset	X	X	
Notes		* Sequencer playback transmission, and Smart Transmit™ in POLY mode.		

Mode 1= OMNI ON, POLY **Mode 2= OMNI ON, MONO** O = YES
Mode 3= OMNI OFF, POLY **Mode 4= OMNI OFF, MONO** X = NO

List of Hints

List of Hints

Latching the Pressure Amount with the Sustain Pedal 5
Scrolling through the Available Files in Flashing LOAD Mode 9
Advance by Screens in Command Mode 9
Advance by Screens in Edit Mode 9
Advancing by Screens Using the Data Entry Controls 10
Resetting Parameter Values with the Cancel•No Button 10
Deleting an Instrument 18
Transmitting MIDI Volume Messages 22
Adjusting Instrument Volumes without Selecting Them 22
Toggling between System and MIDI Edit parameters 35
Assigning Different MIDI Reception Configurations to O.S. Disks 47
Assigning Macro 11 as a Bank File Directory 59
Create Performance Preset Screen 71
Setting Instrument Parameter Values Relative to One Another 75
Direct-Dialing MIDI Program Change Numbers 78
Extending the Pitch Transposition Range 82
Create Performance Preset Screen 86
Viewing Each Instrument's Effect 92
Returning to the ROM Effect from the Bank Effect 93
MIDI Panic button 93
Toggling Between the Mod Destination and the Actual Parameter in an Effect 103
Sampling Fade-Ins and Fade-Outs in Real Time 199
Optimizing WaveSamples 199
Syncing-Up to Other Samples or Sequences During Sampling 201, 218
Recording Samples without a MIDI Keyboard 202
Using the Data Entry Controls to select Audio Tracks 206
Optimizing the Sample Threshold Level 218
Viewing WaveSample Parameter Values by Playing the Keyboard 227
Resetting WaveSample Parameter Values with the Cancel•No button 227
Setting a Modulator OFF 229
Using the ASR–10 with an External Breath Controller (XCTRL) 232
Poly-Key Pressure & Sequencer Memory 233
Copying and Pasting Wave Data Values 239
Resetting LOOP START and END for Synthesized Loop 252
Quickly Verify Time Duration When Using TIME COMPRESS/EXPAND 254
Copying Envelopes 288
Displaying the Current Sequence/Song Screen 306, 329
Using the Pedal•CV to Record Continuous Volume Changes 309
Copy an Instrument to Another Location 310
Displaying the Current Sequence/Song Screen 329
Relocating the sequencer to the last selected GOTO location 329
Setting the Tempo by Tapping on the Enter•Yes button 330
Assigning a MIDI Song Select Number to the Song 336
Setting Track Parameter Values Relative to One Another 339
Transmitting MIDI Volume Messages 340
Using the Pedal•CV to Record Continuous Volume Changes 340
Assigning Different MIDI Reception Configurations to OS Disks 342
Moving a Single Key Event with EVENT EDIT TRACK 352
Transposing a Track by more than one octave 352
Recording Samples without a MIDI Keyboard 383
Optimizing Resampled WaveSamples 384
Creating a "Chorused" Effect 390
Creating Unique Poly-rhythms 410

ASR–10
Index

ASR–10 Index

A

AC Line In 6
ADD DATA 265
Additional Resources xii
ALL NOTES OFF 36
Amp
  Amp Parameters 278
  Cross fade breakpoints 279
    Create a keyboard cross fade 391
Amplifying the ASR–10 vi
Audio Input Specs 6
Audio Input — B/Right and A/Left 6
Audio Track Buttons — A and B 8
Audio Track MIX 343
Audio Track PAN 343
Audio Tracks 324
  Assigning Different Output Routings 419
  Selecting, Enabling, and Disabling Audio Tracks 325
  Using a Guitar with the ASR–10 418
  Using a Microphone with the ASR–10 417
AUDITION page 308
Auto-loop 37
Auxiliary Outputs — Aux 1, Aux 2, Aux 3 6

B

BACKUP/RESTORE 53
Bank
  How to Load a Bank 20
  Loading a Bank 365
  Save Bank 85
  Saving a bank 68, 366
Banks 19, 66
  About Banks 300
BASECHAN PRESSURE 39
BIDIRECTIONAL X-FADE 249
Boost 280
  About the Boost parameter 30
BOWTIE CROSS FADE LOOP 248
Breath Controller 232

C

CD ROM x
CHANGE STORAGE DEVICE 48
Channel Pressure 233
Channel pressure via MIDI 39
CLEAR DATA 262
Click Track Parameters 331
Command/System•MIDI Page 45
Companion Layers 208
COMPUTER 45, 360
CONVERT SAMPLE RATE 253
COPIES WILL CHANGE-OK? 250, 255
COPY DATA 262
COPY FLOPPY DISK 48
COPY LAYER 296
COPY PITCH TABLE 273
COPY SCSI DRIVE 52
COPY TRACK 346
COPY WAVE PARAMETERS 255
Copying Effects 107
CREATE NEW LAYER 296
CREATE NEW SEQUENCE 307
CROSS FADE LOOP 244
CV Pedal x, 3, 36

D

Data Entry Controls 10
Delay Times/Tempo BPM Chart 410
Delete Instrument Effect 87
DELETE LAYER 296
DELETE PITCH TABLE 273
Deleting a File from Disk 29, 371
Deleting an Instrument from the Internal Memory 18
DI-10 Digital I/O Board (S/PDIF) x
Digital I/O — Input/Output 3
Dinosaur 412
Directories
  Create a new directory 48
Disk Capacity — Bytes, Blocks, and Files 359
Disk Drive 10
Disk Error messages 376
Disk Memory 13
Disk Storage 357
Disk Warnings 376
Disks
  Different types of files 358
Display 7
Drum machine 404
  Sequence a Drum Machine from a Track of the ASR–10 404
  Sync a Drum Machine to the ASR–10 404
  Sync the ASR–10 to a Drum Machine 404
Dual Function Effect Mixer 100

E

EDIT PITCH TABLE 271
EDIT SONG STEPS 338
Edit/System•MIDI Page 35
Edit/Wave Parameters 237
Effect
  Delete Instrument Effect 87
EFFECT MOD CONTROL 101
Effect Modulators 102, 103
  List of Effect Mod Sources 102
Effect Parameters
  44KHZ DELAYS 110
  44KHZ REVERB 108
  8-VOICE CHOR 167
  CHOR+REV+DDL 116
  CHORUS+REVRB 111
  CMP+DIST+REV 117
  DE-ESSER 183
  DIST+CHO+REV 119
  DUAL DELAYS 110
  DUCKER 186
  EQ + DELAY LFO 139
  EQ+CHOR+DDL 153
  EQ+CMPRESSOR 175
  EQ+FLNGR+DDL 159
  EQ+TREM+DDL 162
  EQ+VIBR+DDL 156
  EXPANDER 177
  FAST PITCHSH 173
  FLANGER+REV 113
  GATED REVERB 131
  GUITAR AMP 1 145
  GUITAR AMP 2 145
  GUITAR AMP 3 148
  HALL REVERB 108
  HALL REVERB2 121
  INVRS EXPNDR 181
  KEYED EXPANDER 179
  LARGE PLATE 125
  LARGE ROOM 121
  MULTITAP DDL 137
  NLIN REVRB 1 134
  NLIN REVRB 2 134
  NLIN REVRB 3 134
  PARAM EQ 190
  PHASER+DDL 165
  PHASER+REVRB 112
  PITCH SHIFT 169
  PITCH+DDL 171
  REVRSE REVB2 129
  REVRSE REVRB 127
  ROOM REVERB 108
  ROT.SPKR+REV 114
  RUMBLE FILTR 189
  SMALL PLATE 125
  SMALL ROOM 121
  SPKR CABINET 151
  TUNABLE SPKR 152
  VAN DER POL 191
  VCF+DISTORT 142
  WAH+DIST+REV 120
Effect Types
  Bank Effect
    About Bank Effects 92
    To Display the Bank Effect 92
  Instrument Effects
    About Instrument Effects 92
    To Display the Current Effect 92
  ROM Effect
    About ROM Effects 93
    To Display the ROM Effect 93
Effect Variations 108
Effects
  About the Effects i, 91
  Combined Effects 100
  Concepts 91
  Copying the Current Effect 107
  Parameters 107
  Routing Instruments through the Bank Effect 101
  Saving a Bank Effect 107
  Selecting Effects
    List of ROM Effects 94
Effects Busses
  About Busses 99
Effects Mixing 99
ElectroStatic Discharge (ESD) 15
ENSEMBLE CROSS FADE 247
ENTER PLAYS KEY 393
Envelope Parameters 284
Envelopes
  About Envelopes 282
  Copying Envelopes 288
ERASE SONG + ALL SEQS 336
ESD 15
ESP 91
ESP Chip 96
EVENT EDIT TRACK 348
External controller 44
EXTRAPOLATE PITCH TBL 273

F

FADE IN 261
FADE OUT 261
Fadecurves 280

Filter cutoff frequency 277
FILTER EVENT 347
Filters 274
  Cutoff frequency 274
  Filter Parameters 275
  High-pass filter 274
  Low-pass filter 274
  Rolloff curve 274
Floppy Disk
  Copy Floppy Disk 361
  Files 359
  Format a floppy disk 360
  How to Copy the O.S. to Disk 46
  How to Format a Floppy Disk 45
  Two types of formatting 359
Floppy Disks 357
Foot Switch 4
FORMAT SCSI DRIVE 56
FORMAT TYPE 45
FREE DISK BLOCKS 35
FREE SYSTEM BLOCKS 35
Front Panel Controls 7
FX Select/FX Bypass page 99
FX Select•FX Bypass Button 9

G

Glide Mode 292
GLIDE MODES 293
GLOBAL BEND RANGE 36
Global controllers 41, 407
Global effect 92
GOTO Function 329
Ground Loops
  About Ground Loops iv

H

Headphones 8
How to Play Instruments 21

I

Input Level LED Meters 7
Input Level Trim Control 5
Instrument 63
  Copy 83
  Create 83
  Delete 83
  Delete Effect 87
  How to Load an Instrument 16, 64, 363
  Key Range 82
  Name 81
  Save Instrument 84
  Size (in blocks) 80
  What is an Instrument 16
Instrument Configuration 225
Instrument•Sequence Track Buttons 10, 21
Internal Memory 13, 14
INVERT DATA 264

K

Keyboard Range 23
KEYDOWN LAYERS 77
KEYUP LAYERS 77

L

Layer Commands 296
  COPY LAYER 296
  DELETE LAYER 296
Layers
  Layer Parameters 292
LEFT FOOT SW
  Effects Modulation 37
  Foot Switch Functions 37
  Sample Start/Stop 37
  Starting and Stopping the Sequencer 37
Legato layer 293
Level-Detect mode 217
LFO Parameters 289
Line Conditioner iv
Load global parameters 47
Loading a Song or a Single Sequence from Disk 26
Loading an Effect File 97, 370
Loading and Saving Sequencer Data 303, 367
Loading Sequencer Data 26
Loops
  MAKE LOOP LONGER 251
  SYNTHESIZED LOOP 252

M

Macros
  About Macros 48
  Creating a Macro 58
  Invoking a Macro 58
  Loading a Macro File 58
  Saving a Macro File 58
  What Is a Macro? 57
Main audio outputs 6
Main Out — Right/Mono and Left/Mono 6
Making a Stereo Sound from a Mono

WaveSample 389
Manual
About The Manual ii
MASTER TUNE 35
Memory 364
Memory Expansion 13
MERGE TWO TRACKS 348
MERGE WAVESAMPLES 259
Microwave Oven viii
MIDI
About MIDI 38, 397
Controlling Remote Devices 397
Making a MIDI Instrument 399
Recording MIDI Tracks 314
Sequencing MIDI Devices 312
MIDI BASE CHANNEL 38
MIDI Connections v
MIDI CONTROLLERS (ON/OFF) 42
MIDI guitar controller 405
MIDI In 5
MIDI In Mode 40
MIDI Out 5
MIDI OUT CHANNEL 77
MIDI OUT PROGRAM 78
MIDI SONG SELECT (ON/OFF) 43
MIDI Song Selects 409
MIDI STATUS 80
MIDI Sys-Ex recorder 50
Loading Data from Disk 51
Saving Data 50
MIDI SYS-EX RECORDER 42
MIDI Thru 5
MIX WAVESAMPLES 259
Mode Buttons 9
COMMAND mode 9
EDIT mode 9
LOAD mode 9
Modulators
About Modulation 229
Modulation Amount 230
Modulation Sources 230
Selecting a Modulator 229
MONO A Mode 40
MONO B Mode 40
MONO Modes 41
MULTI CONTROLLERS 44
MULTI Mode 40
Multi-Track Digital Recorder 385
Multi-Track Record 315
MultiSampling
How to Multisample 204
Parameters 220

N

NORMALIZE GAIN 256
Nyquist frequency 197

O

OMNI Mode 40
Operating System 12
Optional accessories x
Output 341
Output mixing 100

P

Page Buttons 7
Parameter Illustrations 32
Parametric Programming 11
Patch Select
Making patch select changes 408
Patch Select (Foot Switch) 5
PATCH SELECT MODE 81
Pedal/CV 3
Pedal Specs 4
Performance Preset
Create Preset 86
Creating Keyboard Splits 72
Performance Presets 69
How to Create a Performance Preset 70
Phones 8
Pile o' Instruments 23
Pitch bend range 270
Pitch Commands 271
Pitch Parameters 269
PITCH TABLE 294
Playing a Sequence/Song 27, 306
POLY Mode 40
Poly-Key Pressure
About Poly-Key Pressure 39
Power 7
Power — Polarization and Grounding iii
PRE TRIGGER 219
Pressure (After-touch) 233
PRESSURE MODE 78
Programming Effects 99
Programming Patches 76

Q

Quakercaster 406
QUANTIZE TRACK 345

R

Rear Panel Connections 3
Receiving MIDI Program Changes 42
Receiving Program Changes 43
Recording a Sequence 308
Recording MIXDOWN VOLUME and MIXDOWN PAN 323
Recording Program Changes 314
Recording Song Tracks 321
REPLICATE DATA 263
Resampling through Effects 381
REVERSE CROSS FADE 246
REVERSE DATA 264
Rhythm sequencer 404
Root key 269

S

Sample rate 218
Sample•Source Select Button 8
Sampling
- About Auto-Loop Finding 210
- About Sampling i
- Adjusting the Sampling Threshold Level 200, 217
- Creating a Short Loop 210
- Getting better quality samples. 198
- How to Sample 198
- Long Loops 212
  - Creating a Long Loop 212
- Looping 209
- Multisampling 220
- Sampling Over an Existing WaveSample 205
- Sampling parameters 201, 217
- Setting the Input Level 201
- Short (single-cycle) loop 210
- Short Loop
  - Adjusting Loop Position 211
- Sound
  - What is a sound 195
- Understanding Aliasing 197
- Understanding Sample Rates 196

Sampling Threshold Level 217
Sampling time 219
SAVE GLOBAL PARAMETERS 47
SAVE MACRO FILE 48
SAVE SONG AND ALL SEQS 335
Saving a Single Sequence to Disk 28, 304, 368
Saving a Song (along with all Sequences) to Disk 29, 305, 369
Saving an Effect to Disk 98, 370
Saving an Instrument to Disk 65, 364
Saving and Loading from a SCSI Storage Device 371
Saving Effects 107
SCALE DATA 265
SCALE EVENT 353
SCSI Connector 5
Selecting a Sequence/Song 27
Selecting Effects 94
SEQ REC MODE 332
SEQ REC SOURCE 333
Sequence
- How to string sequences together 27, 306
- Use an external controller to record a track 315, 401

Sequence Tracks 307
Sequencer
- Append Sequence 337
- CHANGE SEQUENCE LENGTH 337
- CLOCK SOURCE 330
- Copy an entire sequence 334
- Create a new sequence 334
- LOOP 330
- Rename the current sequence or song 335
- Save all sequencer memory 335
- Save the current sequence 335
- Tempo 330
- To erase unwanted sequences 334
- Using an External Sequencer 402

Sequencer "Transport Controls" 8
SEQUENCER INFORMATION 336
Sequencer Parameters 329
Sequencer Status 302
Sequencing
- About Digital Sequencing 299
- About Sequencing ii
- About the Sequencer 299

Serial Number
- Where to Find 12

SHIFT TRACKS BY CLOCKS 353
Signal processing 91
SIMM
- Memory allocations 14
- What is a SIMM? 14

SIMM Warning 13
Single Function Effect Mixer 100
Smart Transmit 411
- Pressure 413

Song
- Delete a Step 320
- Insert a step 320

Making a Song 318
Recording Song tracks 321
Song mode 317
Song Position Pointers 409
Song Tracks 321
Sound Libraries x
SPLICE WAVESAMPLES 260
Stacking Instruments 25
Stereo Layer Link 208, 295
Stereo sampling 206
Surge/Spike Suppressor iv
SW-10 Dual Foot Switch x, 4, 37
SW-2 Foot Switch x, 4, 37
SW-6 Foot Switch x, 4
Synthesizer
About The Synthesizer i
System Exclusive 372, 375
Saving Data 372
Storage 372

T

TIME COMPRESS/EXPAND 254
To Send a MIDI Program Change from the ASR–10 42
Track
How to Change the Default Track MIX Volume 309, 340
To erase a selected track 346
Track PAN 341
Track status 339
Tracks
Recording Program Changes 314
"Transport Controls" 301
TRANSPOSE TRACK 352
Triple Function Effect Mixer 101

U

Using an SQ–R PLUS 32 Voice 399

V

Velocity 393
Volume Slider 8
VOLUME SMOOTHING 257

W

WaveSample
About WaveSamples 225
COPY WAVESAMPLE 242
CREATE NEW WAVESAMPLE 242
DELETE WAVESAMPLE 242
Editing 227
Key range 270
Loop end point 239
Loop position 240
Loop start point 239
Creating a loop 239
Modulation Amount and Range 241
Sample end point 238
Sample start point 238
TRUNCATE WAVESAMPLE 243
Wave modulation 240
WaveSample Commands 228
WaveSample Commands in Stereo Layers 228
WaveSample Configuration Chart 226
WAVESAMPLE INFORMATION 243
WaveSample Pan 278
WaveSample playback modes 237
WaveSample Volume 278
Wee-little keyboard 10
What is a Sequence? 300
What is a Song? 300
WRITE DISK LABEL 51
How to Label a Disk 51

X

XPOS OCT - SEMI (Transpose) 82

"INSTRUCTIONS PERTAINING TO A RISK OF FIRE, ELECTRIC SHOCK, OR INJURY TO PERSONS"

IMPORTANT SAFETY INSTRUCTIONS

WARNING—When using electric products, basic precautions should always be followed, including the following:

1. Read all the instructions before using the product.

2. Do not use this product near water - for example, near a bathtub, washbowl, kitchen sink, in a wet basement, or near a swimming pool, or the like.

3. This product should be used only with a cart or stand that is recommended by the manufacturer.

4. This product, either alone or in combination with an amplifier and headphones or speakers, may be capable of producing sound levels that could cause permanent hearing loss. Do not operate for a long period of time at a high volume level or at a level that is uncomfortable. If you experience any hearing loss or ringing in the ears, you should consult an audiologist.

5. The product should be located so that its location or position does not interfere with its proper ventilation.

6. The product should be located away from heat sources such as radiators, heat registers, or other products that produce heat.

7. The product should be connected to a power supply only of the type described in the operating instructions or as marked on the product.

8. This product may be equipped with a polarized line plug (one blade wider than the other). This is a safety feature. If you are unable to insert the plug into the outlet, contact an electrician to replace your obsolete outlet. Do not defeat the safety purpose of the plug.

9. The power supply cord of the product should be unplugged from the outlet when left unused for a long period of time.

10. Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.

11. The product should be serviced by qualified service personnel when:

    a. The power supply cord or the plug has been damaged; or
    b. Objects have fallen, or liquid has been spilled into the product; or
    c. The product has been exposed to rain; or
    d. The product does not appear to operate normally or exhibits a marked change in performance; or
    e. The product has been dropped, or the enclosure damaged.

12. Do not attempt to service the product beyond that described in the user-maintenance instructions. All other servicing should be referred to qualified service personnel.

SAVE THESE INSTRUCTIONS